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This is a request for filing a PROVISIONAL APPLICATION FOR PATENT under 37 CFR 1.53(c).

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TITLE OF THE INVENTION (280 characters max)

OFDM BASED VOICE SERVICE BASED ON ADAPTIVE CHANNEL-AWARE SCHEDULING

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Respectfully submitted,

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OFDM BASED VOICE SERVICE BASED ON ADAPTIVE CHANNEL-AWARE SCHEDULING

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OFDM BASED VOICE SERVICE BASED ON ADAPTIVE CHANNEL-AWARE SCHEDULING

Background of the Invention

[0001] Orthogonal frequency division multiplexing (OFDM) is a special case of multi-carrier transmission, where a single data stream is transmitted over a number of low rate sub-carriers [1]. The main reason to use OFDM for transmission is to enhance the robustness against frequency selective fading or narrow band interference. In a single carrier system, a single fade or interference can cause the entire link to fail, while in a multi-carriers system, only a small percentage of the sub-carriers may be affected. Error correction coding can then be used to correct for the few erroneous sub-carriers.

[0002] Recently, OFDM is widely researched and developed for wireless data transmission in IEEE standards of 802.11 [2] and 802.16 [3]. These standards provide the efficient solutions for high data rate transmission on a broadband OFDM system. Besides the data service, voice service is also an important issue in the future network. However, OFDM is not beneficial to the voice service due to the following reasons. OFDM technique is different from code division multiple accesses (CDMA). CDMA may simply provide a Walsh code to each individual user jointly with power control and easily achieve real-time voice service by transmitting circuit packets. OFDM, however, is hard to perform the power control for individual OFDM tone and only control the number of OFDM tones for each user. This results in a problem how to deal with a complicated frequency channel assignment for users if a constant data rate has to be maintained.

[0003] To deal with the above problem associated to a voice service achievement based on OFDM technique in this memo, we propose the real-time transmission method using channel-aware scheduler with adaptive CIR margin on the forward-link. The channel-aware scheduler is to adaptively control and efficiently deliver the voice packet in every transmission time interval (TTI). The channel-aware scheduler optimizes the OFDM tone assignment for each user while minimizing the number of OFDM tones used for voice transmission. The adaptive CIR margin is used to determine

adaptive modulation and coding (AMC) and significantly reduce the system outage.

Description of the Invention

[0004] The embodiments set forth below represent the necessary information to enable those skilled in the art to practice the invention and illustrate the best mode of practicing the invention. Upon reading the following description in light of the accompanying drawing figures, those skilled in the art will understand the concepts of the invention and will recognize applications of these concepts not particularly addressed herein. It should be understood that these concepts and applications fall within the scope of the disclosure.

Criterion

[0005] Criterion of voice service is concerned with the voice capacity achieved with a certain system outage [4]. System outage for voice users is to be evaluated based on the percentage of voice users in per-link outage. To do this, we need to evaluate the short-term frame error rate (FER) for a voice link by measuring the FER over windows of 400 ms (20 20-ms frames). Per-user outage is defined as the event where a user's voice connection in either direction has short-term FER higher than 15% more often than 1% of the time. This is

$$\text{(Per-user outage for user i)} = \frac{\sum_{j=1}^{M} I_{i,j}}{M} > 1$$

where, M is the simulation run in frame, and the indication function $I_{i,j}$ is defined by

$$I_{i,j} = \begin{cases} 1 & \text{if } FER_{i,j} \geq 15\% \\ 0 & \text{if } FER_{i,j} < 15\% \end{cases}.$$

Thus, the system outage is given by

System outage = Prob.(Per-user outage among all users in all runs), which should be no more than 3%.

Existing Technology for Real-Time Service

[0006] To achieve real-time services including voice and real-time video, we may transmit either circuit packets on a dedicated channel or data packets on a shared channel. The former is easy to be implemented, using single Walsh code for each user jointly with a power control, for example in 1xRTT and 1xEV-DV. The latter is fairly complicated, using a scheduler to assign a shared CDMA channel to user slot by slot, for example in 1xEV-DV. The scheduler could utilize different modulation and code depending on user channel conditions.

[0007] Since OFDM is widely used by exploiting a shared channel for voice and data services, we focus on the discussion associated with scheduler and AMC. With respect to transmission techniques on a shared channel, two important key issues can be raised; one is to employ an efficient scheduler and the other is to employ a proper CIR margin.

- The conventional scheduler we consider for real-time transmission is the round robin[1]. This scheduler cyclically allocates a TTI to each one of the users in order to transmit real-time packet without consideration of the packet quality (successful or not) and channel conditions.
- The CIR margin for AMC determination is a constant margin without consideration of channel conditions.

On the OFDM channel, however, the round-robin scheduler is not efficient way to extract the user diversity gain for real time and constant rate service. Also, the constant CIR margin can help to reduce somewhat user outage, but is not suitable for voice service because it does not take into account voice criterion.

OFDM Based Voice Traffic Transmission

[0008] In this section, we first show the OFDM frame structure hierarchy. Based on the frame structure then, we propose a channel-aware scheduler, which optimizes the OFDM tone assignment for each user while minimizing the number of used tones according to either the instant reported CIR or the average CIR. The scheduler takes the user who experiences the minimum

[1] It should be noted that instead of using round-robin scheduler for voice transmission, a proportional fairness scheduler is used for data transmission.

CIR among the served users. To reduce the system outage, we employ the adaptive CIR margin in consideration of channel conditions and current service quality.

OFDM Frame Structure Hierarchy

[0009] On the forward-link, each arrival voice packet must be transmitted in each OFDM frame with 20 msec time duration. Each OFDM frame consists of 10 TTIs each with the interval of 2 msec. The voice packet can be assigned in any TTI or TTIs depending on the channel conditions[2]. Each TTI is formed by 6 blocks, each having two OFDM symbols. The OFDM symbol includes both preamble part and data part. The detailed OFDM frame structure hierarchy is illustrated in Figure 1.

Channel-Aware Schedulers

[0010] Channel-aware scheduling algorithm takes two steps. The first step is to optimally pre-assign the frequency channels (OFDM tones) to each user in consideration of the channel condition and voice target rate. Then the second step is to properly select an active user for efficient packet transmission. The scheduling is performed every TTI based on the reported CIR averaged over TTI time interval.

Optimum OFDM Tone Assignment

[0011] OFDM channel is concerned with two-dimensional channels: frequency and time as shown in Figure 2. F_m and B_n indicate the OFDM tone number and time block number, respectively. OFDM channels are TTI unit channels where the scheduler is operated for all candidate users.

To carry out the channel-aware assignment for voice service, we require the reported CIRs for all tones[3]. The reported CIR is calculated by averaging instant received CIRs over all blocks in each TTI at receiver. This results in a reasonable CIR reporting interval equivalent to TTI. Due to the feedback signaling, a certain delay for CIR reporting has to be taken into account.

In order to explain the optimum channel assignment, we make the following definitions:

[2] If the user experiences a good channel, one TTI is enough to complete the transmission, otherwise, the voice packet has to be broken down to multiple sub-packets and delivered by multiple TTIs.

[3] In practice, we only need to report the average CIR over 6 blocks for a certain number of OFDM tones and the remained tones could be calculated by using a linear interpolation.

1. There are *N* available OFDM tones (channels) each with the estimated CIR. We sort the *k*-th user in ascending order in terms of their reported CIR, $\Gamma_{n,k}$, $n = 1,2,\ldots,N$. It is noted that the maximum number of available OFDM tones for each TTI could be equal to $N_F \times N_B$, where N_F and N_B represent the number of OFDM tones and the number of time blocks, respectively.
2. The voice packet transmission allows to use repetitions due to bad channel conditions. That means if $N_{U,k}$ tones are assigned for user *k*, we only use $M_{R,k}$ tones for the voice packet transmission and the remained tones are used for repetition to enhance the received CIR. The relationship between $N_{U,k}$ and $M_{R,k}$ is $N_{U,k} = M_{R,k} \cdot m_k + M_{C,k}$ as illustrated in Figure 3.
3. The voice payload transmitted in each frame should be kept constant with a target R_{TARGET}.
4. The minimum payload for each TTI transmission is required to be R_{MIN}. That means, the scheduler never allows to transmit the packet whose payload size is less than R_{MIN}.

[0012] The maximum number of assigned tones are limited with N_{MAX}. This happens when the reported CIRs for all tones are extremely low and the repetition transmission is dominated on the voice packet.

The optimization of OFDM tone assignment for the *k*-th user is achieved with the minimization of the total used tones while keeping the payload to be slightly larger than or equal to the predetermined payload in each voice frame. Thus, the optimization problem for OFDM tone assignment has the following form

$$\min_{M_{R,k}} \quad N_{U,k}$$

$$\text{subject to} \quad \overline{\Gamma}_k(N_{U,k}, M_{R,k}) = \frac{1}{M_{R,k}} \cdot \sum_{n=1}^{N_{U,k}} \Gamma_{n,k}$$

$$f(\overline{\Gamma}_k) = R_k$$

$$M_{R,k} \cdot R \geq R_{\text{MIN}}$$

$$\begin{cases} M_{R,k} \cdot R > R_{\text{TARGET}} & \text{if } N_{U,k} < N_{\text{MAX}} \\ M_{R,k} \cdot R \le R_{\text{TARGET}} & \text{if } N_{U,k} = N_{\text{MAX}} \end{cases}$$

for $N_{U,k} = 1,2,\ldots,N$ and $M_{R,k} = 1,2,\ldots,N_{U,k}$

where $f(\cdot)$ is the mapping function of data rate (equivalent to payload transmitted in each TTI) obtained from the link level simulation results.

Implementation of Optimum OFDM Tone Assignment

[0013] The optimization problem shown above is a nonlinear formula that may be simplified thanks to the mapping function from the link level curve. According to the discussion in Appendix-I, we see that the number for $M_{R,k}$ should be taken as large as possible so as to minimize the number of used OFDM tones $N_{U,k}$.

[0014] Before discussing the algorithm, we make the following two definitions:

1. *N* is the number of remained OFDM tones after assigning a certain number of active users.
2. $N_{\text{MIN},k}$ is the minimum number of OFDM tones required for user *k* transmission due to the minimum CIR limitation, defined as

$$N_{\text{MIN},k} = \frac{\text{CIR threshold for the worst MCS link level curve}}{\text{Maximum report CIR over all tones for user } k}$$

The operation of optimum OFDM tone assignment is individually performed TTI by TTI for each user. The procedure of algorithm for optimum OFDM tone assignment is described as follows:

1. Check the candidate users which require transmitting the voice signal. The candidate user is defined as a user, who is waiting for either original packet transmission or retransmission.
    a. If there are some users, go to step-2.
    b. Otherwise, terminate the operation.
2. Sort the reported CIRs for user *k* in all available OFDM tones in ascending order.
3. Determine the minimum number of OFDM tones $N_{\text{MIN},k}$ required for user *k* transmission.

4. Compare the number of remained tones with the minimum number of tones $M_{\mathrm{MIN},k}$
   a. If the number of remained OFDM tones *N* is larger than or equal to the minimum number of OFDM tones $M_{\mathrm{MIN},k}$, go to step-5.
   b. Otherwise, go back to step-1.
5. Set $N_{U,k} = M_{\mathrm{MIN},k}$.
6. Set $M_{R,k} = N_{U,k}$.
7. Calculate the average CIR based on given $M_{R,k}$ and $N_{U,k}$. Map the CIR into data rate curve (equivalent to payload).
   a. If the mapped data rate is larger than or equal to the predetermined target data rate, put the assigned OFDM tone indices into the memory and terminate the operation for the current user, and then go back to step-1.
   b. Otherwise, go to step-8.
8. Subtract one from $M_{R,k}$, i.e., $M_{R,k} = M_{R,k} - 1$.
9. Determine if $M_{R,k}$ is equal to 0.
   a. If $M_{R,k} = 0$, go to step-10.
   b. Otherwise, go back to step-7.
10. Add one into $N_{U,k}$, i.e., $N_{U,k} = N_{U,k} + 1$.
11. Compare $N_{U,k}$ with the maximum number of OFDM tones $N_{\max}$.
    a. If $N_{U,k} > N_{\mathrm{MAX}}$, put the assigned OFDM tone indices into the memory and go back to step-1.
    b. Otherwise, go to step-6.

The detailed flow-chart of optimum OFDM tone assignment for all users is shown in Figure 4.

<u>Active User Selection</u>

[0015] The adaptive scheduler is designed for determining the active user based on the optimum OFDM tone assignment as discussed above. We define two the scheduling factors for determining the active users: one is the

minimum reported CIR $\Gamma_{\mathrm{MIN},k}$ for user k among assigned OFDM tones, and the other is the average CIR $\bar{\Gamma}_k$ for user k over assigned OFDM tones.

The procedure of algorithm for the user scheduling is described as follows:

1. Call the module for optimum OFDM tone assignment for all candidate users (see Figure 4).
2. Determine the scheduling factor for all candidate users: either minimum reported CIR $\Gamma_{\mathrm{MIN},k}$ or average CIR $\bar{\Gamma}_k$.
3. Select a user with the minimum factor as an active user.
4. Remove the OFDM tones which are already assigned for scheduled active user and indicate unused OFDM tones for next user scheduling.
5. Check the remained OFDM tones N for next user scheduling.
   a. If a certain number of OFDM tones are available, go to step-6.
   b. Otherwise, end up the scheduling operation.
6. Check the remained user needed for voice transmission.
   a. If there are some users, go back to step-1.
   b. Otherwise, end up the scheduling operation.

The detailed flow-chart of adaptive scheduling for all users is illustrated in Figure 5.

<u>An Example of Scheduling</u>

[0016] As we discussed above, active user scheduling in each TTI takes two operations; one is to assign the optimum OFDM tones to each candidate user, and the other is to select the active user who experiences the minimum scheduling factor.

[0017] To make this algorithm clearly understanding, we give a simple example in consideration of 3 candidate users, 8 OFDM tones for each symbol and 6 blocks for each TTI. The example for the user scheduling in each TTI is illustrated in Figure 6, where 'o' indicates the assigned tones for candidate users, and 'x' indicates the occupied tones for active users.

The procedure of scheduling is described as follows:

1. For the first user scheduling, all OFDM tones (6×8 channels) are available. Using the optimum OFDM tone assignment and the channel indication of $\mathrm{CH}_{f,b}$ (*f* and *b* indicate the tone and block indexes, respectively), we have

a. For user #1, assigned channels are $CH_{2,1}$, $CH_{2,2}$, $CH_{2,3}$, $CH_{2,4}$, and $CH_{2,5}$.

b. For user #2, assigned channels are $CH_{3,1}$, $CH_{3,2}$, $CH_{3,3}$, $CH_{3,4}$, $CH_{3,5}$, $CH_{3,6}$, $CH_{2,1}$, and $CH_{2,2}$.

c. For user #3, assigned channels are $CH_{6,1}$, $CH_{6,2}$, $CH_{6,3}$, and $CH_{6,4}$.

d. Select user #2 as the first active user based on the minimum scheduling factor.

2. For the second user scheduling, the OFDM tones $CH_{3,1}$, $CH_{3,2}$, $CH_{3,3}$, $CH_{3,4}$, $CH_{3,5}$, $CH_{3,6}$, $CH_{2,1}$, and $CH_{2,2}$ are already occupied by the active user #2, and the remained tones are available for next two user scheduling. Using the optimum OFDM tone assignment again, we have

    a. For user #1, assigned channels are $CH_{2,3}$, $CH_{2,4}$, $CH_{2,5}$, $CH_{2,6}$, $CH_{1,1}$, and $CH_{1,2}$.

    b. For user #3, assigned channels are $CH_{6,1}$, $CH_{6,2}$, $CH_{6,3}$, and $CH_{6,4}$.

    c. Select user #1 as the second active user based on the minimum scheduling factor.

3. For the third user scheduling, the OFDM tones $CH_{3,1}$, $CH_{3,2}$, $CH_{3,3}$, $CH_{3,4}$, $CH_{3,5}$, $CH_{3,6}$, $CH_{2,1}$, $CH_{2,2}$, $CH_{2,3}$, $CH_{2,4}$, $CH_{2,5}$, $CH_{2,6}$, $CH_{1,1}$, and $CH_{1,2}$ are already occupied by the active user #1 and #2, and the remained tones are available for next user scheduling. Using the optimum OFDM tone assignment again, we have

    a. For user #3, assigned channels are $CH_{6,1}$, $CH_{6,2}$, $CH_{6,3}$, and $CH_{6,4}$.

    b. Select user #3 as the third active user.

4. In this case, all the users become the active users. The OFDM tones used for user packet transmission are

    $CH_{3,1}$ $CH_{3,2}$ $CH_{3,3}$ $CH_{3,4}$ $CH_{3,5}$ $CH_{3,6}$

$CH_{2,1}$ $CH_{2,2}$ $CH_{2,3}$ $CH_{2,4}$ $CH_{2,5}$ $CH_{2,6}$

$CH_{1,1}$ $CH_{1,2}$ $CH_{6,1}$ $CH_{6,2}$ $CH_{6,3}$ $CH_{6,4}$

and the remained tones will be assigned for data transmission.

This scheduling operation is repeated TTI by TTI.

Practical OFDM Channel Assignment

[0018] Signaling between transmitter and receiver for the optimum OFDM channel assignment could be very complicated if the total number of OFDM tones is large, for instance, 729 tones in Nortel Mid-Term project [6]. To reduce the signaling complexities, we consider a unit for channel assignment called *unit channel*. The unit channel consists of multiple OFDM tones and multiple blocks, resulting in a two dimensional channel assigned for voice transmission. Each user packet could be transmitted on single unit channel or multiple unit channels depending on the channel conditions. The unit channel may be formed with any structure as long as it can be easily predetermined between transmitter and receiver for signaling. Figure 7 and Figure 8 illustrate examples of unit channels. In the first example, each unit channel consists of 12 neighbor OFDM tones on both frequency and time domains. In the second example, there are eight unit channels, each formed by many OFDM tones that are spread into entire frequency band and time blocks.

[0019] In case of high user velocity with high Doppler frequency, the channel changes fast in time domain. On such a channel, we need to form a unit channel by spreading the OFDM tones in entire frequency band to achieve frequency diversity for voice transmission. In case of low velocity with low Doppler frequency, however, we need to form a unit channel with neighbor OFDM tones resulting in user diversity due to the accurate channel estimation we may obtain.

Unit Channel Addressing

[0020] To inform the unit channel address to each user, it is required to transmit the signaling information on control channels in each TTI. Here, we try to figure out the number of bits for maximum and minimum required signaling:

- When we assume that all unit channels N are used for serving all voice users U, the maximum required signaling bits $S_{CH}^{(MAX)}$ can be written as

$$S_{\mathrm{CH}}^{(\mathrm{MAX})} = N \cdot \log_2 N.$$

- When we assume that each user occupies only one unit channel, the minimum required signaling bits $S_{\mathrm{CH}}^{(\mathrm{MIN})}$ can be written as

$$S_{\mathrm{CH}}^{(\mathrm{MIN})} = U \cdot \log_2 N.$$

If N = 256 and U = 40, for example, the number of bits for maximum and minimum required signaling are $S_{\mathrm{CH}}^{(\mathrm{MAX})} = 2048$ bits and $S_{\mathrm{CH}}^{(\mathrm{MIN})} = 320$ bits, respectively. Practically, the number of required signaling bits should be between $S_{\mathrm{CH}}^{(\mathrm{MIN})}$ and $S_{\mathrm{CH}}^{(\mathrm{MAX})}$.

[0021] In order to reduce the required signaling amount, we consider multiple unit-channel addressing method. This method takes different size for unit channels depending on the channel where the user delivers voice packet. That means, if the user experiences a low CIR, it is efficient to utilize the unit channels with larger size, and otherwise with smaller size.

An example for multiple unit channels with total 64 units is illustrated in Figure 9. This example shows four different unit channels assigned for different CIR users.

- Unit channel-1 provides the *smallest* unit channel (also called basic unit channel) with total unit channels of 64. This channel is used to transmit the voice packet for the user who experiences a *very high* CIR.
- Unit channel-2 provides the *small* unit channel with total unit channels of 32. This channel is used to transmit the voice packet for the user who experiences a *high* CIR.
- Unit channel-3 provides the *large* unit channel with total unit channels of 16. This channel is used to transmit the voice packet for the user who experiences a *low* CIR.
- Unit channel-4 provides the *largest* unit channel with total unit channels of 8. This channel is used to transmit the voice packet for the user who experiences a *very low* CIR.

[0022] In this example, it requires two signaling bits to indicate the unit channel size and six bits to indicate the unit channel address for specific user. For example, if the user requires 32 basic unit channels for voice packet

transmission, Table 1 lists the different required signaling bits by using different unit channels.

Table 1: Comparison between different unit channel used for voice transmission.

Used Unit Channel	Required Signaling Bits
1	$8 \times 32 = 256$
2	$8 \times 16 = 128$
3	$8 \times 8 = 64$
4	$8 \times 4 = 32$

From this table, it can be seen that using unit channel-4 significantly reduces the number of signaling bits. The reducing factor could be eight by compared to unit channel-1.

[0023] Moreover, a user could have multi-unit channels depending on how many basic unit channels are used. For instance, if the user requires 34 basic unit channels for voice packet transmission, we may have 4 largest unit channels (unit channel-4) and 1 small unit channel (unit channel-2).

In addition, the following things have to be taken into account if the multiple unit channel approach is employed for indicating the channel address:

- The number of unit channels N_{UCH} must be 2^n for $n = 0,1,2,\Lambda$.
- Since the number of control channels is pre-limited, the scheduler has to look at how many control channels currently remains and then assigns the active user.
- The total signaling bits to indicate each OFDM unit channel should be $n + \log_2 N$.

CIR Margin for Adaptive Modulation and Coding

[0024] Due to the delay constraint for voice service, the transmission or retransmission must be accomplished within a short time frame as compared to data transmission. This requires a robustness of retransmission if the original transmission fails or robustness of transmission if current short-term FE within 400 msec is high. To do this, we consider two kinds of flexible CIR margins: one depends on the number of retransmissions within a frame duration (20msec) and the other depends on the number of frame error (FE) over windows of 400msec.

Retransmission Based Adaptive Margin

[0025] The retransmission based adaptive margin depends on the number of retransmissions within frame duration of 20 msec. That means, the margin value increases as the number of retransmissions increases. The adaptive margin $\Delta\left(n_k^{(\mathrm{ReTx})}\right)$ for user *k* can be represented as

$$\Delta\left(n_k^{(\mathrm{ReTx})}\right) = f_{\mathrm{Margin}}\left(n_k^{(\mathrm{ReTx})}\right)$$

where $n_k^{(\mathrm{ReTx})}$ is the number of retransmissions for user-*k*, and $f_{\mathrm{Margin}}(\cdot)$ is the margin function which could be either linear or concave increment function.

Frame Error Based Adaptive Margin

[0026] Frame error based adaptive margin depends on the number of frame error (FE) over windows of 400 msec intervals. That means, the margin value increases as the number of frame error increases. The adaptive margin $\Delta\left(n_k^{(FE)}\right)$ can be represented as

$$\Delta\left(n_k^{(\mathrm{FE})}\right) = \begin{cases} f_{\mathrm{Margin}}\left(n_k^{(\mathrm{FE})}\right) & \text{if } n_k^{(\mathrm{FE})} < \eta \\ 0 & \text{Otherwise} \end{cases}$$

where $n_k^{(\mathrm{FE})}$ is the cumulated number of frame errors updated every 400 msec window duration, and $f_{\mathrm{Margin}}(\cdot)$ is the margin function which could be either linear or concave increment function. In addition, η is the number frame errors allowed to happen over 400 msec window without any impact of voice performance, practically, $\eta = 3$ (see the voice criterion as discussed above).

System Level Performance

[0027] In order to investigate the voice performance with the proposed schemes based on OFDM technique, we perform the system level simulation. In the following, we first describe the simulation assumptions as well as channel model. Then, we show the simulation results with detailed discussions.

Simulation Assumptions

[0028] Based on the Nortel Mid-Term OFDM project [6], the system level simulation assumptions are listed in Table 2.

Table 2: Simulation assumptions.

Number of cells	19
Number of users	40, 60, 80, 100 and 120
FFT/IFFF size	1024
Antenna structure	1x1
Frame duration	20 msec
TTI interval	2 msec
Number of blocks in each TTI	6
Number of symbols in each block	2
Maximum retransmission number	5
Average CIR report delay	2 TTI
Target BLER	0.01
Voice Data Rate	9.6 kbps
Soft Handoff	No
Entire bandwidth	6 MHz
Maximum sub-packets for voice packet	5
Maximum number of OFDM tones in each block	729
Maximum number of OFDM tones in each TTI	729*6
Maximum number of tones for each user	729*6/10

The margins we consider for AMC determination is as follows:

- The margin we considered with a constant value is 0dB for simplicity.
- The margin based on retransmission number is given by

$$\Delta\left(n_k^{(\mathrm{ReTx})}\right) = n_k^{(\mathrm{ReTx})} \ (\mathrm{dB}).$$

- The margin based on frame error number is given by

$$\Delta\left(n_k^{(\mathrm{FE})}\right) = \begin{cases} 0\,(\mathrm{dB}) & \eta = 0 \\ 3\,(\mathrm{dB}) & \eta = 1 \\ 6\,(\mathrm{dB}) & \eta = 2 \\ 0\,(\mathrm{dB}) & \eta \geq 3 \end{cases}.$$

Link-Level Curves for AMC

[0029] The link-level curves in term of block error rate (BLER) are generated according to the Mid-Term project (advanced OFDM system) at WTL of Nortel Networks [6]. The link-level simulation is concerned with 1x1 antenna structure occupying 6MHz bandwidth. Figure 10 shows the link-level code sets in terms of BLER versus SNR.

Channel Model and CIR Generation

[0030] A mixed channel model with assignment probability as listed in Table 3 is considered in our system level simulation.

Table 3: Channel Model.

Channel Model	# of fingers	Speed (km/h)	Fading	Assignment Probability
Model A	6	3	SCM	0.3
Model B	6	10	SCM	0.3
Model C	6	30	SCM	0.2
Model D	6	120	SCM	0.1
Model E	6	0, f_D=1.5Hz	SCM	0.1

The fading channel generation is based on a spatial channel model (SCM) [7]. The CIR calculation is performed on the frequency domain and individually generated for each OFDM tone. The methodology is based on CIR calculation at Mid-Term project [8].

Performance Analysis

[0031] For the sake of simplicity, we give the short names for the schedulers as follows:

- Scheduler-1: the scheduler chooses the user who experiences the lowest CIR among all users. The CIR for each user is determined by picking up the lowest reported CIR among all active tones.
- Scheduler-2: the scheduler chooses the user who experiences the lowest CIR among all users. The CIR for each user is determined by averaging reported CIRs over all active tones.
- Scheduler-3: the round-robin scheduler.

We also give the short names for the margins as follows:

- Margin-1: the retransmission based adaptive margin.
- Margin-2: the frame error based adaptive margin.
- Margin-3: the constant margin.

[0032] The following is the investigations of the system level performances in terms of system outage whereby we may figure out the system capacity for OFDM based voice service, and the number of remained OFDM tones whereby we may figure out how much channel resource is used for voice transmission. Based on the provided simulation result, we may further discuss the proposal and find out the advantages and disadvantages.

Simulation Results without Consideration of Channel Signaling

[0033] In this subsection, we do not take into account the signaling information for OFDM sub-carrier indication. In this case, thus, the unit channel is formed by only one sub-carrier.

[0034] Figure 11 shows the system outage as a function of the number of served users when we exploit the margin-2 for AMC determination for various specified schedulers. From the figure, we may make several observations as follows:

1. When the number of users increases, the system outage increases as well.
2. The system outage performance is slightly dependent on the schedulers although the scheduler-2 provides the best performance.

The voice capacities calculated by the system outage that is not allowed to be more than 3% are 58, 60 and 43 for scheduler-1, 2, and 3, respectively.

[0035] Figure 12 shows the system outage as a function of the number of served users when we exploit the scheduler-2 for various specified margins. It can be seen that the system outage performance is strongly dependent on the type of margin. That is, Margin-2 provides the best results while Margin-3 provides the worst results.

[0036] Figure 13 shows the average number of remained OFDM tones as a function of served users when we exploit margin-2 for AMC determination for various specified schedulers. We observe that the number of remained OFDM tones decreases when the number of users increases. Also, the number of remained OFDM tones is almost independent of what kind of

scheduler is used although the scheduler-3 occupies slightly more OFDM tones for voice transmission.

[0037] Figure 14 shows the average number of remained OFDM tones as a function of served users and Figure 15 shows the CDF of the number of remained OFDM tones when we exploit the scheduler-2 for various specified margins. It can be observed that the number of remained OFDM tones depends upon what kind of margin is used. That is, Margin-1 occupies more OFDM tones than others.

[0038] Finally, we investigate the system level performance in terms of system outage and the number of remained OFDM tones if we do not employ the optimum OFDM tone assignment, as listed Table 4 and Table 5, respectively.

Table 4: System outage without optimum OFDM tone assignment for 60 served users.

	Margin-1	Margin-2	Margin-3
Scheduler-1	0.067	0.125	0.092
Scheduler-2	0.162	0.164	0.148
Scheduler-3	0.033	0.083	0.037

Table 5: Average remained OFDM tones without optimum OFDM tone assignment for 60 served users.

	Margin-1	Margin-2	Margin-3
Scheduler-1	1602.2	1609.2	1649.8
Scheduler-2	1292.8	1240.4	1344.1
Scheduler-3	2685.6	2714.7	2733.7

The performance in terms of system outage is almost identical with the optimum OFDM tone assignment. However, it requires much more OFDM tones to meet the same target.

We make a comparison of system outage, remained tones for specific values of number of users (from 40 to 120) as listed in Table 6. It can be seen that by compared to the capacity of 40 users in 1xRTT or 1xEV-DV with the bandwidth of 1.25 MHz, the OFDM based voice service achieves the same user capacity but only occupies 12% OFDM channels (equivalent to used bandwidth of 0.72 MHz). The frequency efficiency is more than 1.70.

Table 6: Comparison for the user capacity when we employ Scheduler-2 and Margin-2.

Number of Users	40	60	80	100	120
System Outage	0.013	0.030	0.067	0.103	0.122
Remained Tones	3849	3683	3454	3232	2827
Occupied Channels (%)	12.0	15.8	21.0	26.1	35.3
Equivalent Bandwidth (MHz)	0.72	0.95	1.26	1.57	2.12

<u>Simulation Results with Consideration of Channel Signaling</u>

[0039] In this subsection, we take into account the signaling transmission. This signaling transmission performed from base-station to mobile on the control channel could be very large amount data. For comparison, we take following considerations:

1. Two types of unit channel for voice packet delivery; one is formed by 6 OFDM tones and the other by 18 OFDM tones.
2. Two types of CIR margin;

$$\Delta\left(n_k^{(\mathrm{FE})}\right)=\begin{cases}0\,(\mathrm{dB}) & \eta=0\\ 3\,(\mathrm{dB}) & \eta=1\\ 6\,(\mathrm{dB}) & \eta=2\\ 0\,(\mathrm{dB}) & \eta\geq 3\end{cases} \quad \text{or} \quad \Delta\left(n_k^{(\mathrm{FE})}\right)=\begin{cases}0\,(\mathrm{dB}) & \eta=0\\ 5\,(\mathrm{dB}) & \eta=1\\ 10\,(\mathrm{dB}) & \eta=2\\ 0\,(\mathrm{dB}) & \eta\geq 3\end{cases}.$$

3. Active users are assigned by scheduler-2.
4. The number of voice users is 40 or 60.

Table 7 and Table 8 list the system outage rate for voice service when the number of users is 40 and 60, respectively. From the tables, we may observe

that the system outage rate can meet the voice requirement if the CIR margin is 0, 5, 10 dB.

Table 7: System outage rate of voice service when the number of users is 40.

	Unit Channel (OFDM Tones)	
CIR Margin (dB)	6	18
0,3,6	0.138	0.085
0,5,10	0.038	0.025

Table 8: System outage rate of voice service when the number of users is 60.

	Unit Channel (OFDM Tones)	
CIR Margin (dB)	6	18
0,3,6	0.117	0.100
0,5,10	0.025	0.058

Table 9 and Table 10 list the remained OFDM tones and equivalent bandwidth BW_{EQ} when the number of users is 40 and 60, respectively. From the tables, we may see that by compared to the bandwidth required by 1xRTT or 1xEV-DV, the bandwidth occupied by the OFDM based voice users is still small if the unit channel is formed by 6 OFDM tones but is large if by 18 OFDM tones.

Table 9: Remained OFDM Tones and equivalent bandwidth when the number of users is 40.

	Unit Channel (OFDM Tones)	
CIR Margin (dB)	6	18
0,3,6	3545(Tones)/1.14(MHz)	3283(Tones)/1.49(MHz)
0,5,10	3525(Tones)/1.16(MHz)	3245(Tones)/1.55(MHz)

Table 10: Remained OFDM Tones and equivalent bandwidth when the number of users is 60.

	Unit Channel (OFDM Tones)	
CIR Margin (dB)	6	18
0,3,6	3363(Tones)/1.39(MHz)	2901(Tones)/2.02(MHz)
0,5,10	3330(Tones)/1.43(MHz)	2816(Tones)/2.14(MHz)

[0040] Next, we investigate how much signaling data is required to indicate the unit channel address. Since the number of unit channels N_{UCH} must be 2^n for $n = 0,1,2,\Lambda$, we give several cases with N_{UCH} = 2, 4, 8 and 16. To meet the voice outage requirement, moreover, we decide to use the CIR margin with 0,5,10 dB.

Table 11: Required average signaling bits when the number of voice users is 40.

	Number of Unit Channels			
Unit Channel Size (Tones)	2	4	8	16
6	808	316	210	226
18	294	142	123	134

Table 12: Required Average signaling bits when the number of voice user is 60.

	Number of Unit Channels			
Unit Channel Size (Tones)	2	4	8	16
6	991	406	299	328
18	409	202	178	196

[0041] Table 11 and Table 12 list the required average signaling amount in bits in each TTI when the number of voice users is 40 and 60, respectively. By considering the minimum signaling amount as a criterion, we may conclude that the best choice for the number of unit channels is 8.

[0042] Figure 16 shows the CDF of signaling bits for the indication of the unit channel address when the number of users is 40, the CIR margin is 0-5-

10, and the unit channel size is 18, for various specified values of the number of unit channels, 2, 4, 8 and 16.

[0043] In order to compare the equivalent bandwidth $BW_{\mathrm{EQ}}^{(\mathrm{TOTAL})}$ with 1xRTT or 1xEV-DV, we consider the following formula can be considered

$$BW_{\mathrm{EQ}}^{(\mathrm{TOTAL})} = BW_{\mathrm{EQ}} \cdot \left(1 + \frac{Q_{\mathrm{S}}}{Q_{\mathrm{D}}}\right)$$

where BW_{EQ} is the equivalent bandwidth without consideration of unit channel signaling, and Q_{S} and Q_{D}, respectively, are the signaling bits required to indicate the OFDM unit channel and the transmitted voice packet bits both in frame with 20 msec intervals. And Q_{S} and Q_{D} can be represented as

Q_{S} = (Average Signaling Bits) × (Number of TTIs in Each Frame)

and

Q_{D} = (Voice Transmission Rate) × (Frame Interval) × (Number of Users).

[0044] Based on the above equations and the average signaling bits in Table 11, we may figure out the equivalent bandwidth $BW_{\mathrm{EQ}}^{(\mathrm{TOTAL})}$ for 40 voice users, as listed in Table 13.

Table 13: Equivalent bandwidth for OFDM based voice transmission with consideration of the signaling for OFDM unit channel indication.

	Number of Unit Channels			
Unit Channel Size (Tones)	2	4	8	16
6	2.39	1.64	1.48	1.51
18	2.14	1.84	1.80	1.82

From this comparison, we may find that the practical OFDM based voice transmission requires an additional 18.7% bandwidth as opposed to 1xRTT or 1xEV-DV.

[0045] In terms of voice packet retransmission, so far, we assume that the scheduler could select different OFDM tone channels as compared to the first initial transmission. This results in an additional signaling data for

retransmission. To further reduce the signaling between transmitter and receiver, it can be reasonably assumed that the OFDM channels used for retransmission are the same as the first initial transmission used.

[0046] Figure 17 shows the probability of number of voice packet retransmissions when the number of users is 40, CIR margin is 0-5-10dB, and the number of unit channels is 8. From this result, we may figure out that the percentage of retransmission for unit channel size of 6 is 43.4%.

[0047] By using above percentage number, we then recalculate the equivalent bandwidth used for OFDM based voice transmission, and find that the equivalent bandwidth is 1.30MHz, which is only 4.2% more bandwidth as compared to 1xRTT or 1xEV-DV.

<u>Conclusions</u>

[0048] This application has discussed a new issue associated with OFDM based voice transmission. We have proposed the real-time transmission method using channel-aware scheduler with adaptive CIR margin particularly for voice transmission on the forward-link. By sufficiently utilizing the scheduler in consideration of the minimum CIR factor which the user experiences and adaptively controlling the frame error based margin for AMC determination, we may achieve a very high voice capacity with a low system outage. To reduce the signaling transmission, we have considered multiple unit-channel addressing method. Based on the system level simulation results we discussed above, we may make the following conclusions:

1. Voice capacity associated with system outage depends strongly upon the types of margin. The frame error based adaptive margin (Margin-2) is the best approach to reduce the system outage by increasing the margin value. For instance, increasing the margin value from 3dB and 6dB to more, we may have less system outage and more the user capacity. It is noted that increasing the margin value may increase the occupied OFDM tones for voice transmission.
2. The performances in terms of system outage and the number of remained OFDM tones are slightly dependent on the type of scheduler. This is because in our simulation the data transmission is dominant due to a wide frequency bandwidth and a small number of users (see Table 6) and the voice users have enough tones to be

chosen for transmission. If OFDM has to be used for voice transmission on a narrow frequency bandwidth channel, the schedulers we have proposed above become more and more important.

3. Optimum OFDM tone assignment is very important technique especially for OFDM system. This assignment can be simply implemented, and significantly reduces the number of OFDM tones for voice transmission. This may guarantee the high data rate transmission for data application and achieve an OFDM based DV system.
4. If we consider the same OFDM channels used for retransmission as initial transmission used, the signaling data between transmitter and receiver can be significantly reduced. The equivalent bandwidth is 1.30MHz, which is only 4.2% more bandwidth as compared to 1xRTT or 1xEV-DV.
5. By compared to the capacity of 40 users in 1xRTT or 1xEV-DV with the bandwidth of 1.25 MHz, the OFDM based voice service achieves the same user capacity but occupies the equivalent bandwidth of 1.48 MHz which has 18.7% more bandwidth than 1xRTT. This frequency efficiency loss is due to the soft handoff which OFDM based voice transmission does not use.
6. With respect to the complexity, the optimum OFDM tone assignment requires a complicated signaling. To reduce the signal complexity, we need a unit channel, but it may cause some degradation in performance.

<u>Appendix-I</u>

[0049] Figure 18 shows the transmission rate as a function of CIR threshold for MCS as comparison between real curve and approximation. It can be seen that both curves match well as a convex function and the function can be approximated as a formula

$$f(\Gamma) = \alpha \cdot \Gamma^{\beta} \qquad \text{for } \alpha = 5.5 \text{ and } \beta = 0.42 .$$

Now, the question we raise is whether we need to use repetition or not. To answer this question, we have to look at the property of convex function.

According to the proposition of convex [5], if f_1 and f_2 are both convex functions on the convex set, the function $f_1 + f_2$ is convex on the same set. And it must satisfy

$$f(\Gamma) = \alpha \cdot \left(\sum_{l=1}^{L} \Gamma_l \right)^{\beta} \le \sum_{l=1}^{L} f(\Gamma_l) = \alpha \cdot \sum_{l=1}^{L} \Gamma_l^{\beta} \qquad \text{where } \Gamma = \sum_{l=1}^{L} \Gamma_l .$$

Therefore, we can see that to maximize the transmission rate, we have to choose L as large as possible. For practical reason, however, we may not set L to be a large number because for any transmission the minimum CIR is required. This results in having a maximum number for L.

[0050] The following description provides an overview of a wireless communication environment and the architecture of a base station, or like access point, and a mobile terminal.

[0051] With reference to Figure 19, a base station controller (BSC) 10 controls wireless communications within multiple cells 12, which are served by corresponding base stations (BS) 14. In general, each base station 14 facilitates communications using OFDM with mobile terminals 16, which are within the cell 12 associated with the corresponding base station 14. The movement of the mobile terminals 16 in relation to the base stations 14 results in significant fluctuation in channel conditions. As illustrated, the base stations 14 and mobile terminals 16 may include multiple antennas to provide spatial diversity for communications.

[0052] A high level overview of the mobile terminals 16 and base stations 14 of the present invention is provided prior to delving into the structural and functional details of the preferred embodiments. With reference to Figure 20, a base station 14 configured according to one embodiment of the present invention is illustrated. The base station 14 generally includes a control system 20, a baseband processor 22, transmit circuitry 24, receive circuitry 26, multiple antennas 28, and a network interface 30. The receive circuitry 26 receives radio frequency signals bearing information from one or more remote transmitters provided by mobile terminals 16 (illustrated in Figure 4). Preferably, a low noise amplifier and a filter (not shown) cooperate to amplify and remove broadband interference from the signal for processing.

Downconversion and digitization circuitry (not shown) will then downconvert the filtered, received signal to an intermediate or baseband frequency signal, which is then digitized into one or more digital streams.

[0053] The baseband processor 22 processes the digitized received signal to extract the information or data bits conveyed in the received signal. This processing typically comprises demodulation, decoding, and error correction operations. As such, the baseband processor 22 is generally implemented in one or more digital signal processors (DSPs) or application-specific integrated circuits (ASICs). The received information is then sent across a wireless network via the network interface 30 or transmitted to another mobile terminal 16 serviced by the base station 14.

[0054] On the transmit side, the baseband processor 22 receives digitized data, which may represent voice, data, or control information, from the network interface 30 under the control of control system 20, and encodes the data for transmission. The encoded data is output to the transmit circuitry 24, where it is modulated by a carrier signal having a desired transmit frequency or frequencies. A power amplifier (not shown) will amplify the modulated carrier signal to a level appropriate for transmission, and deliver the modulated carrier signal to the antennas 28 through a matching network (not shown). Modulation and processing details are described in greater detail below.

[0055] With reference to Figure 21, a mobile terminal 16 configured according to one embodiment of the present invention is illustrated. Similarly to the base station 14, the mobile terminal 16 will include a control system 32, a baseband processor 34, transmit circuitry 36, receive circuitry 38, multiple antennas 40, and user interface circuitry 42. The receive circuitry 38 receives radio frequency signals bearing information from one or more base stations 14. Preferably, a low noise amplifier and a filter (not shown) cooperate to amplify and remove broadband interference from the signal for processing. Downconversion and digitization circuitry (not shown) will then downconvert the filtered, received signal to an intermediate or baseband frequency signal, which is then digitized into one or more digital streams.

[0056] The baseband processor 34 processes the digitized received signal to extract the information or data bits conveyed in the received signal. This

processing typically comprises demodulation, decoding, and error correction operations, as will be discussed on greater detail below. The baseband processor 34 is generally implemented in one or more digital signal processors (DSPs) and application specific integrated circuits (ASICs).

[0057] For transmission, the baseband processor 34 receives digitized data, which may represent voice, data, or control information, from the control system 32, which it encodes for transmission. The encoded data is output to the transmit circuitry 36, where it is used by a modulator to modulate a carrier signal that is at a desired transmit frequency or frequencies. A power amplifier (not shown) will amplify the modulated carrier signal to a level appropriate for transmission, and deliver the modulated carrier signal to the antennas 40 through a matching network (not shown). Various modulation and processing techniques available to those skilled in the art are applicable to the present invention.

[0058] In OFDM modulation, the transmission band is divided into multiple, orthogonal carrier waves. Each carrier wave is modulated according to the digital data to be transmitted. Because OFDM divides the transmission band into multiple carriers, the bandwidth per carrier decreases and the modulation time per carrier increases. Since the multiple carriers are transmitted in parallel, the transmission rate for the digital data, or symbols, on any given carrier is lower than when a single carrier is used.

[0059] OFDM modulation requires the performance of an Inverse Fast Fourier Transform (IFFT) on the information to be transmitted. For demodulation, the performance of a Fast Fourier Transform (FFT) on the received signal is required to recover the transmitted information. In practice, the IFFT and FFT are provided by digital signal processing carrying out an Inverse Discrete Fourier Transform (IDFT) and Discrete Fourier Transform (DFT), respectively. Accordingly, the characterizing feature of OFDM modulation is that orthogonal carrier waves are generated for multiple bands within a transmission channel. The modulated signals are digital signals having a relatively low transmission rate and capable of staying within their respective bands. The individual carrier waves are not modulated directly by the digital signals. Instead, all carrier waves are modulated at once by IFFT processing.

[0060] In the preferred embodiment, OFDM is used for at least the downlink transmission from the base stations 14 to the mobile terminals 16. Each base station 14 is equipped with *n* transmit antennas 28, and each mobile terminal 16 is equipped with *m* receive antennas 40. Notably, the respective antennas can be used for reception and transmission using appropriate duplexers or switches and are so labeled only for clarity.

[0061] With reference to Figure 22, a logical OFDM transmission architecture is provided according to one embodiment. Initially, the base station controller 10 will send data to be transmitted to various mobile terminals 16 to the base station 14. The base station 14 may use the CQIs associated with the mobile terminals to schedule the data for transmission as well as select appropriate coding and modulation for transmitting the scheduled data. The CQIs may be directly from the mobile terminals 16 or determined at the base station 14 based on information provided by the mobile terminals 16. In either case, the CQI for each mobile terminal 16 is a function of the degree to which the channel amplitude (or response) varies across the OFDM frequency band.

[0062] The scheduled data 44, which is a stream of bits, is scrambled in a manner reducing the peak-to-average power ratio associated with the data using data scrambling logic 46. A cyclic redundancy check (CRC) for the scrambled data is determined and appended to the scrambled data using CRC adding logic 48. Next, channel coding is performed using channel encoder logic 50 to effectively add redundancy to the data to facilitate recovery and error correction at the mobile terminal 16. Again, the channel coding for a particular mobile terminal 16 is based on the CQI. The channel encoder logic 50 uses known Turbo encoding techniques in one embodiment. The encoded data is then processed by rate matching logic 52 to compensate for the data expansion associated with encoding.

[0063] Bit interleaver logic 54 systematically reorders the bits in the encoded data to minimize the loss of consecutive data bits. The resultant data bits are systematically mapped into corresponding symbols depending on the chosen baseband modulation by mapping logic 56. Preferably, Quadrature Amplitude Modulation (QAM) or Quadrature Phase Shift Key (QPSK) modulation is used. The degree of modulation is preferably chosen

based on the CQI for the particular mobile terminal. The symbols may be systematically reordered to further bolster the immunity of the transmitted signal to periodic data loss caused by frequency selective fading using symbol interleaver logic 58.

[0064] At this point, groups of bits have been mapped into symbols representing locations in an amplitude and phase constellation. When spatial diversity is desired, blocks of symbols are then processed by space-time block code (STC) encoder logic 60, which modifies the symbols in a fashion making the transmitted signals more resistant to interference and more readily decoded at a mobile terminal 16. The STC encoder logic 60 will process the incoming symbols and provide *n* outputs corresponding to the number of transmit antennas 28 for the base station 14. The control system 20 and/or baseband processor 22 will provide a mapping control signal to control STC encoding. At this point, assume the symbols for the *n* outputs are representative of the data to be transmitted and capable of being recovered by the mobile terminal 16. See A.F. Naguib, N. Seshadri, and A.R. Calderbank, "Applications of space-time codes and interference suppression for high capacity and high data rate wireless systems," Thirty-Second Asilomar Conference on Signals, Systems & Computers, Volume 2, pp. 1803-1810, 1998, which is incorporated herein by reference in its entirety.

[0065] For the present example, assume the base station 14 has two antennas 28 (n=2) and the STC encoder logic 60 provides two output streams of symbols. Accordingly, each of the symbol streams output by the STC encoder logic 60 is sent to a corresponding IFFT processor 62, illustrated separately for ease of understanding. Those skilled in the art will recognize that one or more processors may be used to provide such digital signal processing, alone or in combination with other processing described herein. The IFFT processors 62 will preferably operate on the respective symbols to provide an inverse Fourier Transform. The output of the IFFT processors 62 provides symbols in the time domain. The time domain symbols are grouped into frames, which are associated with a prefix by like insertion logic 64. Each of the resultant signals is up-converted in the digital domain to an intermediate frequency and converted to an analog signal via the corresponding digital up-conversion (DUC) and digital-to-analog (D/A) conversion circuitry 66. The

resultant (analog) signals are then simultaneously modulated at the desired RF frequency, amplified, and transmitted via the RF circuitry 68 and antennas 28. Notably, pilot signals known by the intended mobile terminal 16 are scattered among the sub-carriers. The mobile terminal 16, which is discussed in detail below, will use the pilot signals for channel estimation.

[0066] Reference is now made to Figure 23 to illustrate reception of the transmitted signals by a mobile terminal 16. Upon arrival of the transmitted signals at each of the antennas 40 of the mobile terminal 16, the respective signals are demodulated and amplified by corresponding RF circuitry 70. For the sake of conciseness and clarity, only one of the two receive paths is described and illustrated in detail. Analog-to-digital (A/D) converter and down-conversion circuitry 72 digitizes and downconverts the analog signal for digital processing. The resultant digitized signal may be used by automatic gain control circuitry (AGC) 74 to control the gain of the amplifiers in the RF circuitry 70 based on the received signal level.

[0067] Initially, the digitized signal is provided to synchronization logic 76, which includes coarse synchronization logic 78, which buffers several OFDM symbols and calculates an auto-correlation between the two successive OFDM symbols. A resultant time index corresponding to the maximum of the correlation result determines a fine synchronization search window, which is used by fine synchronization logic 80 to determine a precise framing starting position based on the headers. The output of the fine synchronization logic 80 facilitates frame acquisition by frame alignment logic 84. Proper framing alignment is important so that subsequent FFT processing provides an accurate conversion from the time to the frequency domain. The fine synchronization algorithm is based on the correlation between the received pilot signals carried by the headers and a local copy of the known pilot data. Once frame alignment acquisition occurs, the prefix of the OFDM symbol is removed with prefix removal logic 86 and resultant samples are sent to frequency offset correction logic 88, which compensates for the system frequency offset caused by the unmatched local oscillators in the transmitter and the receiver. Preferably, the synchronization logic 76 includes frequency offset and clock estimation logic 82, which is based on the headers to help

estimate such effects on the transmitted signal and provide those estimations to the correction logic 88 to properly process OFDM symbols.

[0068] At this point, the OFDM symbols in the time domain are ready for conversion to the frequency domain using FFT processing logic 90. The results are frequency domain symbols, which are sent to processing logic 92. The processing logic 92 extracts the scattered pilot signal using scattered pilot extraction logic 94, determines a channel estimate based on the extracted pilot signal using channel estimation logic 96, and provides channel responses for all sub-carriers using channel reconstruction logic 98. In order to determine a channel response for each of the sub-carriers, the pilot signal is essentially multiple pilot symbols that are scattered among the data symbols throughout the OFDM sub-carriers in a known pattern in both time and frequency. Continuing with Figure 5, the processing logic compares the received pilot symbols with the pilot symbols that are expected in certain sub-carriers at certain times to determine a channel response for the sub-carriers in which pilot symbols were transmitted. The results are interpolated to estimate a channel response for most, if not all, of the remaining sub-carriers for which pilot symbols were not provided. The actual and interpolated channel responses are used to estimate an overall channel response, which includes the channel responses for most, if not all, of the sub-carriers in the OFDM channel.

[0069] The frequency domain symbols and channel reconstruction information, which are derived from the channel responses for each receive path are provided to an STC decoder 100, which provides STC decoding on both received paths to recover the transmitted symbols. The channel reconstruction information provides equalization information to the STC decoder 100 sufficient to remove the effects of the transmission channel when processing the respective frequency domain symbols

[0070] The recovered symbols are placed back in order using symbol de-interleaver logic 102, which corresponds to the symbol interleaver logic 58 of the transmitter. The de-interleaved symbols are then demodulated or de-mapped to a corresponding bitstream using de-mapping logic 104. The bits are then de-interleaved using bit de-interleaver logic 106, which corresponds to the bit interleaver logic 54 of the transmitter architecture. The de-

interleaved bits are then processed by rate de-matching logic 108 and presented to channel decoder logic 110 to recover the initially scrambled data and the CRC checksum. Accordingly, CRC logic 112 removes the CRC checksum, checks the scrambled data in traditional fashion, and provides it to the de-scrambling logic 114 for de-scrambling using the known base station de-scrambling code to recover the originally transmitted data 116.

[0071] Continuing with Figure 5, a relative variation measure may be determined by providing the channel response information from the channel estimation function 96 to a channel variation analysis function 118, which will determine the variation and channel response for each of the sub-carriers in the OFDM frequency band, and if standard deviation is used, calculate the standard deviation associated with the frequency response. As noted, channel gain is a preferred measure of the channel response for calculating a CQI. The channel gain may be quantified based on a relative amplitude of the channel frequency response in decibels (dB), and as such, the amplitude of the channel frequency response may be represented by $H_{dB}(k)$, which is a function of a sub-carrier index k, where $k=1 \ldots k_{MIN}, \ldots k_{MAX}, \ldots k_{FFT}$. Notably, k_{FFT} is the number of sub-carriers in the entire OFDM frequency band, and the sub-carriers k_{MIN} through k_{MAX} represent the sub-carriers within the OFDM frequency band that are actually used to transmit data. Typically, a range of sub-carriers at either end of the range of sub-carriers are not used, in order to minimize interference with other transmissions. As such, the degree of variation of the amplitude of the channel response may be determined only for the range of sub-carriers being used to transmit data (k_{MIN} through k_{MAX}). The standard deviation of the channel response across the usable range of sub-carriers is calculated as follows:

$$std = \sqrt{\frac{1}{N_u - 1} \sum_{k_{MIN}}^{k_{MAX}} \left(H_{dB}(k) - \overline{H}_{dB}\right)^2} ,$$

where N_u is the number of usable sub-carriers, $H_{dB}(k)$ is the log amplitude of the channel frequency response, and $\overline{H}_{db}$ is the mean of the log amplitude of

the channel response across the usable range of sub-carriers or a subset thereof.

[0072] In a multiple-input multiple-output (MIMO) system where there are multiple transmit and multiple receive antennas 28, 40 each link corresponding to transmit/receive antenna pairs will have a unique CQI. An aggregate CQI, or set of aggregate CQIs, may be required for the overall MIMO set of links. To determine the aggregate CQIs, the channel frequency response and CIR for each transmit and receive antenna pair is determined.

[0073] For multiple receive antennas 40, the multiple channel frequency responses are combined, to provide for the diversity achieved from the multiple receive antennas 40. This combining is an averaging of the power of the respective channel frequency responses across the OFDM frequency band. The channel variation measure is then determined across the combined channel frequency response. The CIR values for the respective multiple receive antennas 40 are combined by summing.

[0074] For multiple transmit antennas 28, the modification to the CQI will depend on the particular space time coding technique employed to reflect the method by which the transmit diversity is being achieved by the code and used by the system. Some schemes, such as transmit diversity, will require that the respective channel frequency responses from the multiple transmit antennas 28 be combined as described for the multiple receive antennas 40 by averaging the power of the channel frequency responses across the OFDM frequency band. The channel variation measure is made across the combined frequency response. Further, the CIR values for the multiple transmit antennas 28 are also combined. For other schemes, a separate CQI may be determined for each transmit antenna 28 and relayed back to the base station 14. The base station 14 may use the CQI per transmit antenna 28 to separately adapt the modulation and coding on the data transmitted on the respective transmit antennas 28.

[0075] Once the channel variation analysis is provided, a variation measure is provided to a CQI function 120 or to the baseband processor 34 for transmission back to the base station 14 via the transmit circuitry 36, depending on the configuration of the embodiment. If the CQI is determined at the base station 14, then the mobile terminal 16 will provide information

indicative of the CIR as well as the variation analysis to the base station 14, which will calculate a CQI and control scheduling as well as coding and modulation for subsequent transmissions to the mobile terminal 16. If the CQI is generated at the mobile terminal 16 and transmitted to the base station 14, the CQI function 120 will receive a CIR from a CIR function 122 and will use the CIR and the variation measurement to either calculate or look up through a look-up table an appropriate CQI, which is then transmitted to the base station 14 via the transmit circuitry 36.

[0076] The CIR function 122 will preferably receive channel response information from the channel estimation function 96 and determine the CIR based on the relative strengths of the desired carrier in light of other interferers in traditional fashion. When the pilot symbols are passed through the channel estimation function 96, the pilot symbols are filtered in a manner exploiting the known pilot symbols to remove noise and interference. The output of the channel estimation function 96 is intended to be a noiseless replica of the pilot symbol. With this replica, the carrier power may be determined, as well as subtracted from the received pilot symbol to yield a noise plus interference signal. This resulting signal is computed to provide an interference power, which is compared to the carrier power to determine the CIR. One example of determining a CIR is provided in co-assigned U.S. patent application serial number 10/038,916 filed January 8, 2002. Those skilled in the art will recognize numerous techniques for determining the CIR. Importantly, the CQI, whether calculated at the mobile terminal 16 or at the base station 14, is based on the variation measure indicia, preferably in light of a CIR. Since the CIR for an OFDM system fails to account for the respective responses for each of the sub-carriers used for transmission, providing a CQI based on the CIR and the variation measure indicia significantly improves the performance of the OFDM system by allowing the base station 14 to better predict an appropriate coding and modulation technique, as well as to provide scheduling among the multiple users.

[0077] The following items are incorporated herein by reference in their entireties:

[1] Richard Van Nee and Ramjee Prasad, *OFDM for Wireless Multimedia Communications*, Artech House, Boston-London, 2000.

[2] IEEE 802.11a, Supplement to IEEE standard for Information Technology, Part-11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications, September 16, 1999.

[3] IEEE 802.16a, IEEE standard for Local and Metropolitan Area Networks, Part-16: Air Interface for Fixed Broadband Wireless Access Systems, April 1, 2003.

[4] 1xEV-DV Evaluation Methodology – Addendum (V6), July 25, 2001.

[5] David G. Luenberger, *Linear and Nonlinear Programming,* Addison-Wesley Publishing Company, Inc., Second Edition, 1984.

[6] Nortel Mid-Term OFDM project, Nortel Networks, WTL, 2002.

[7] Spatial Channel Model Text Description, SCM Text V6.0, April 22. 2003.

[8] J. Wu, "Methodology of CIR Calculation for OFDM System", Mid-Term project at Nortel Networks, April 14, 2003.

[0078] Those skilled in the art will recognize improvements and modifications to the preferred embodiments of the present invention. All such improvements and modifications are considered within the scope of the concepts disclosed herein and the claims that follow.

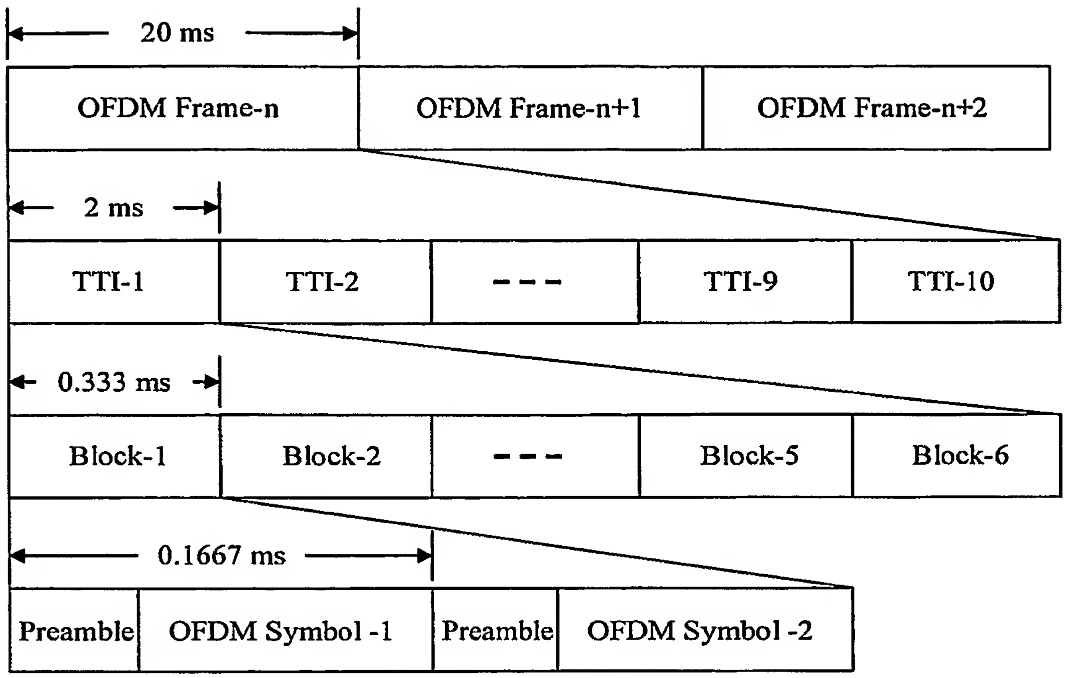

Figure 1: OFDM frame structure hierarchy.

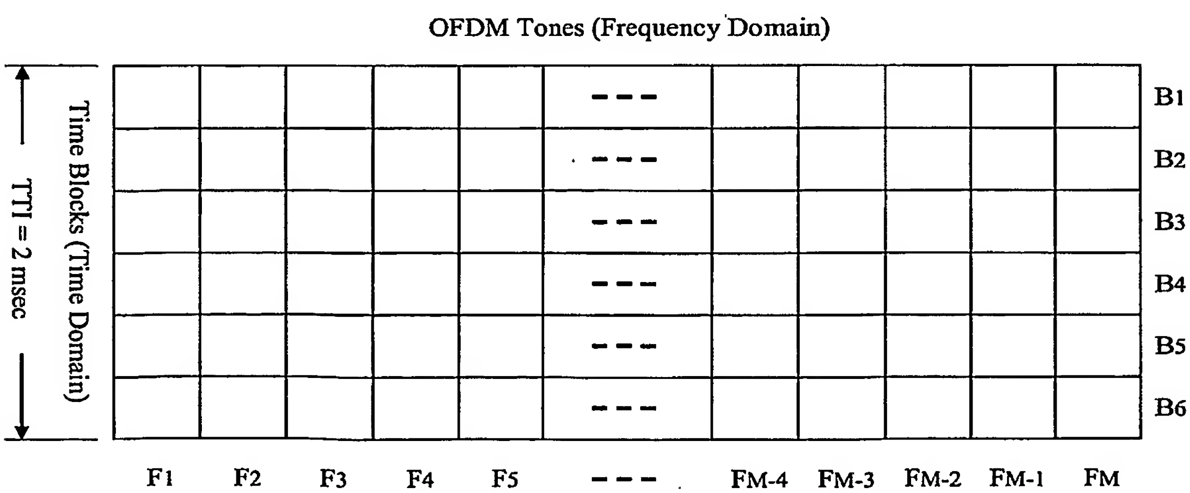

Figure 2: Two-dimensional channels in each TTI.

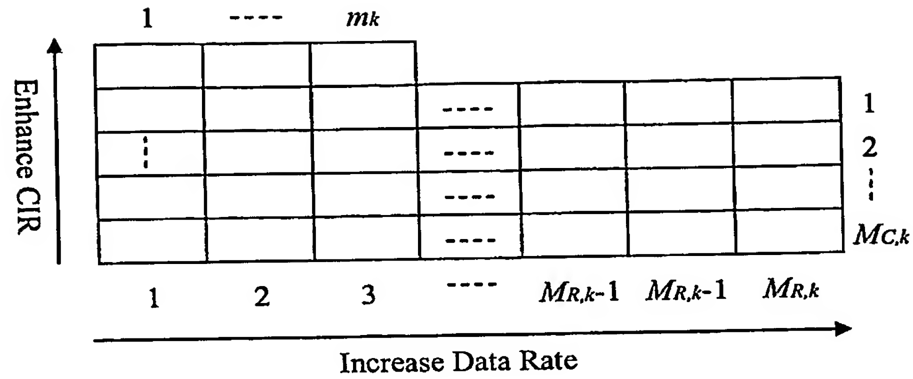

Figure 3: Assignment for OFDM tones with transmission repetition.

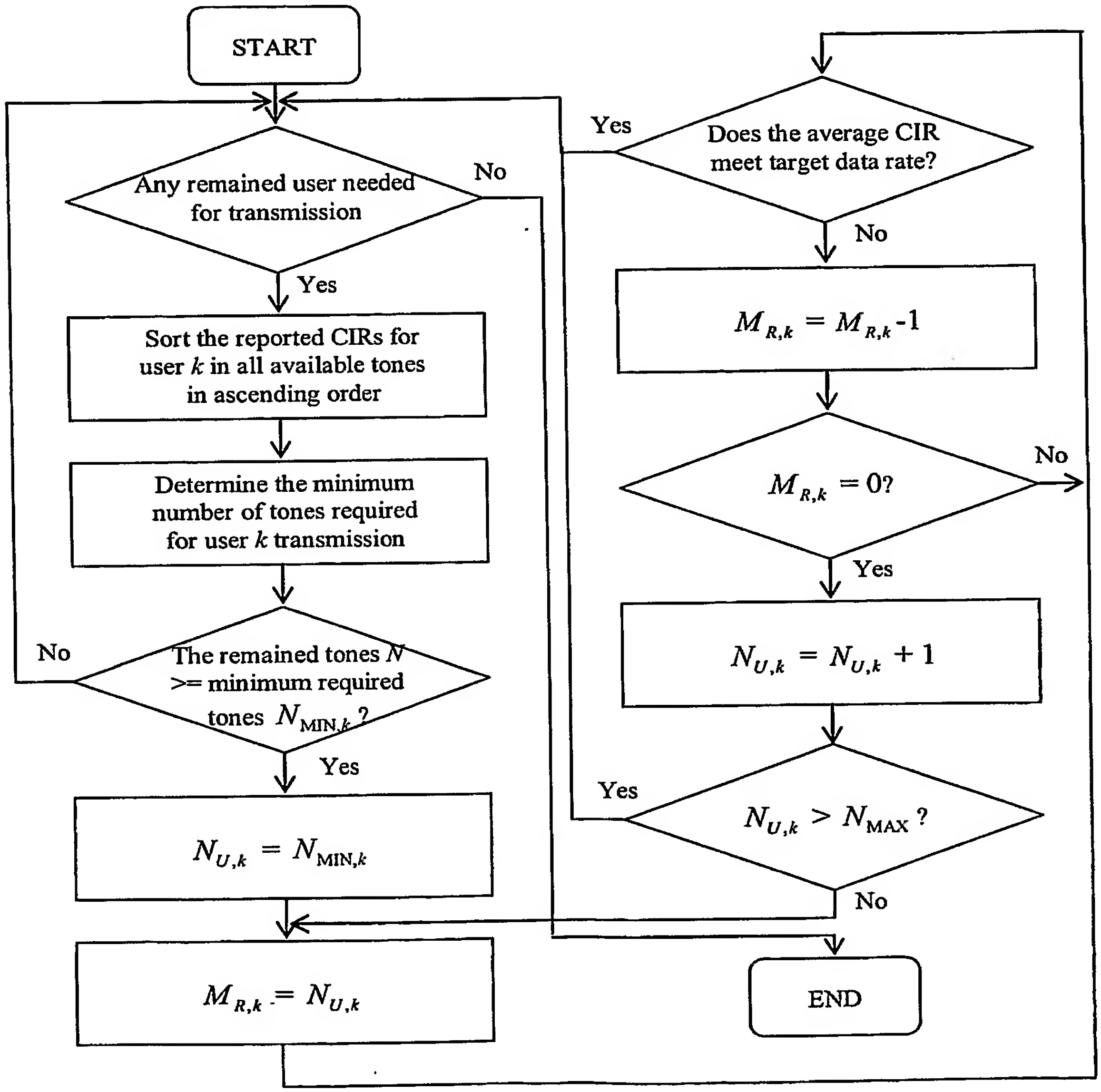

Figure 4: Flow-chart of optimum OFDM tone assignment for all users.

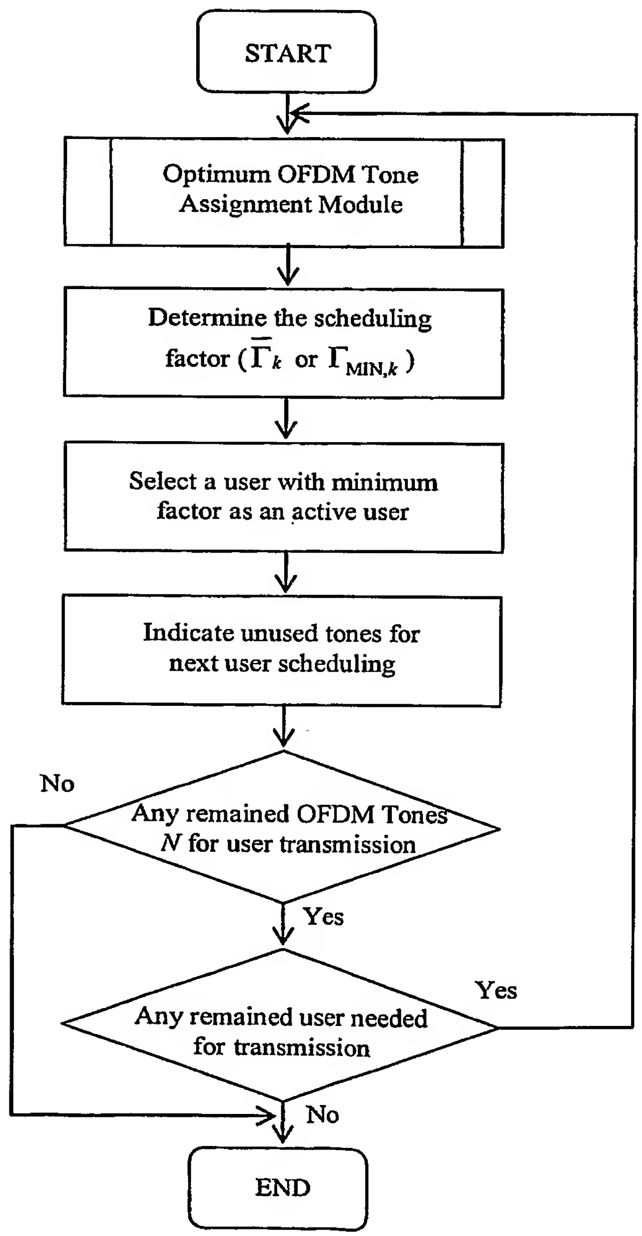
START
Optimum OFDM Tone Assignment Module
Determine the scheduling factor ($\overline{\Gamma}_k$ or $\Gamma_{MIN,k}$)
Select a user with minimum factor as an active user
Indicate unused tones for next user scheduling
Any remained OFDM Tones N for user transmission
No
Yes
Any remained user needed for transmission
Yes
No
END

Figure 5: Flow-chart of adaptive user scheduling.

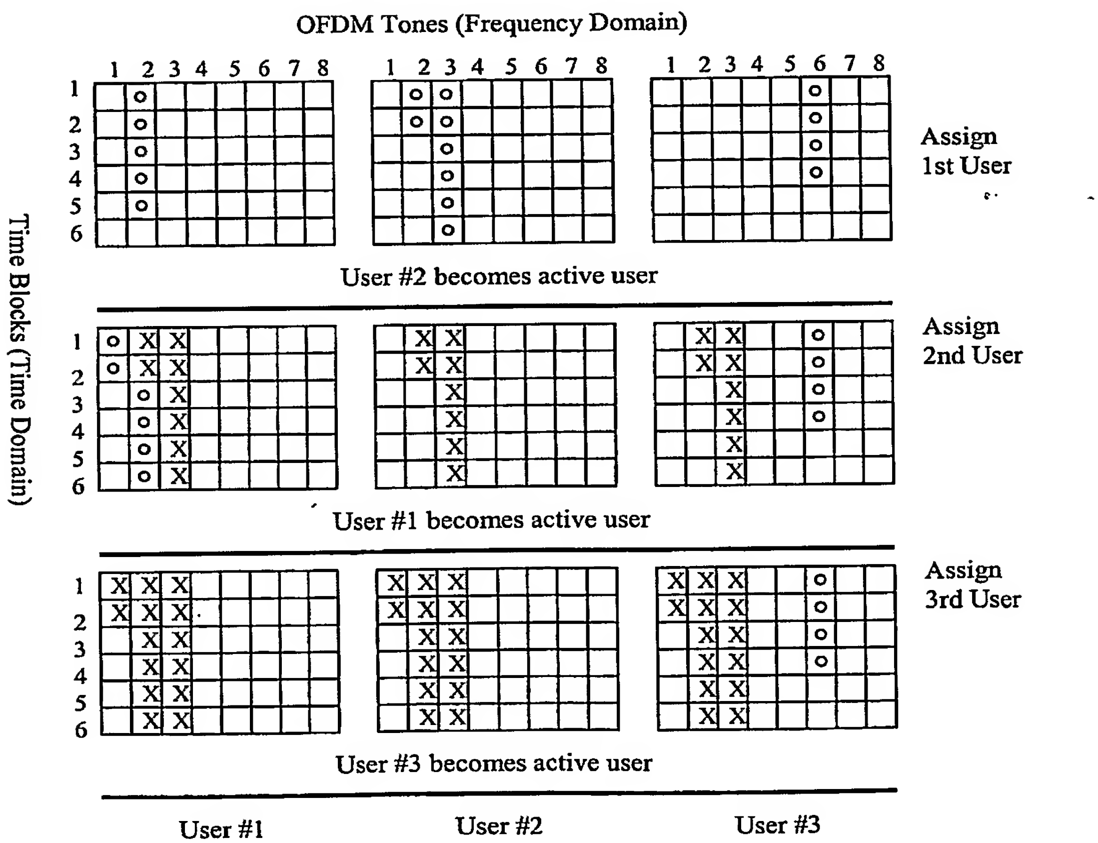

Figure 6: An example of scheduling with 3 candidate users, 8 OFDM tones for each symbol and 6 blocks for each TTI.

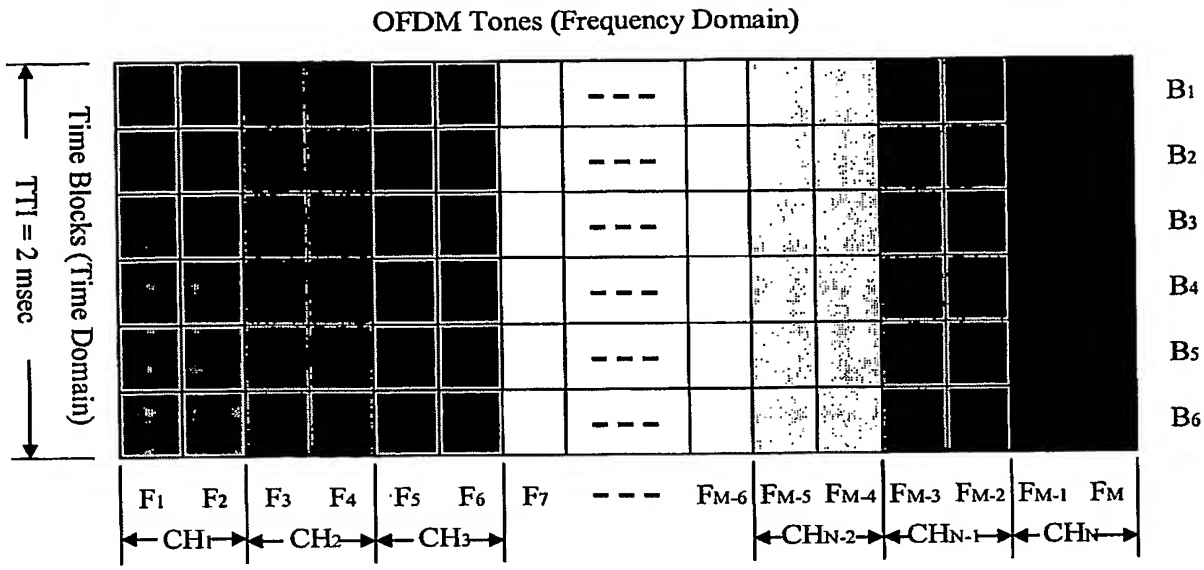

Figure 7: An example of unit channel structure for OFDM system, each consisting of 12 neighbor OFDM tones on both frequency and time domain.

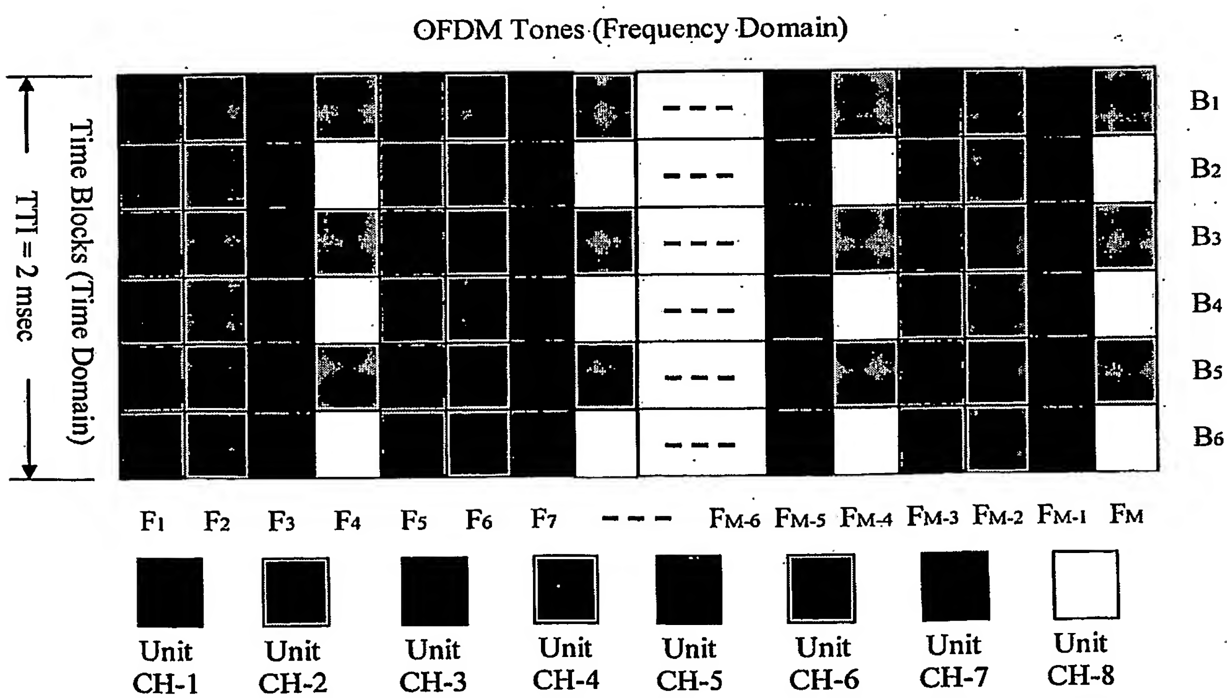

Figure 8: An example of unit channel structure for OFDM system, each consisting of 12 neighbor OFDM tones on both frequency and time domain.

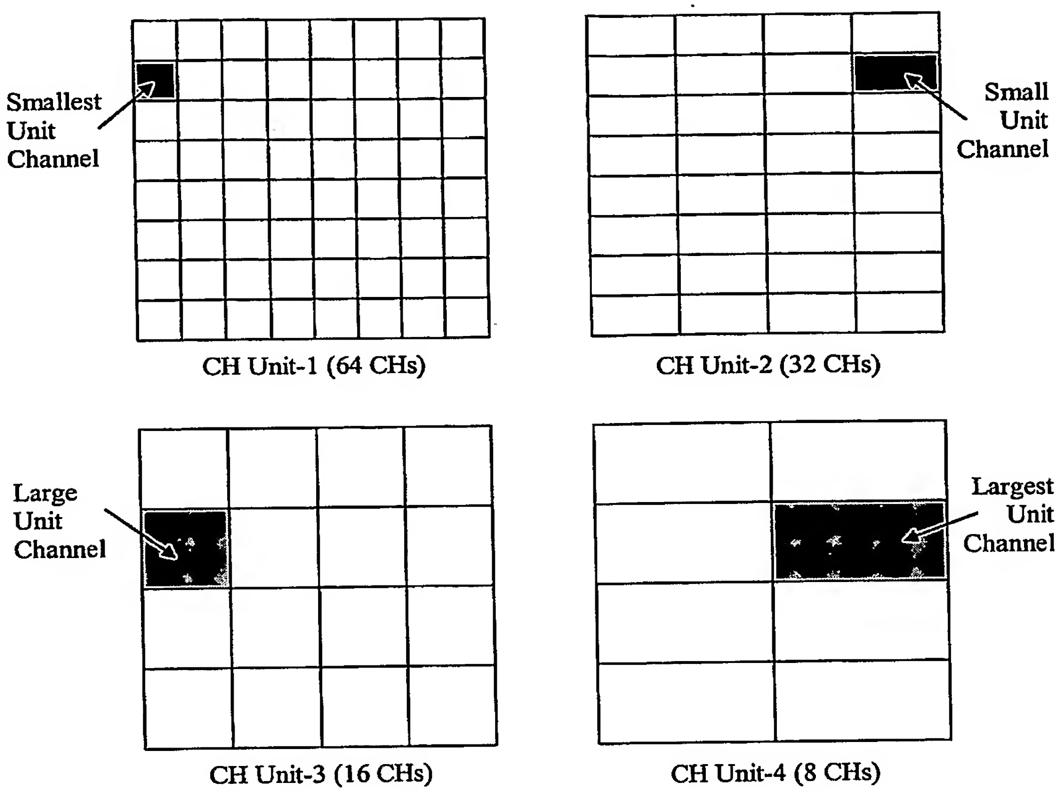

Figure 9: An example of multiple channel units for reducing the signaling channels.

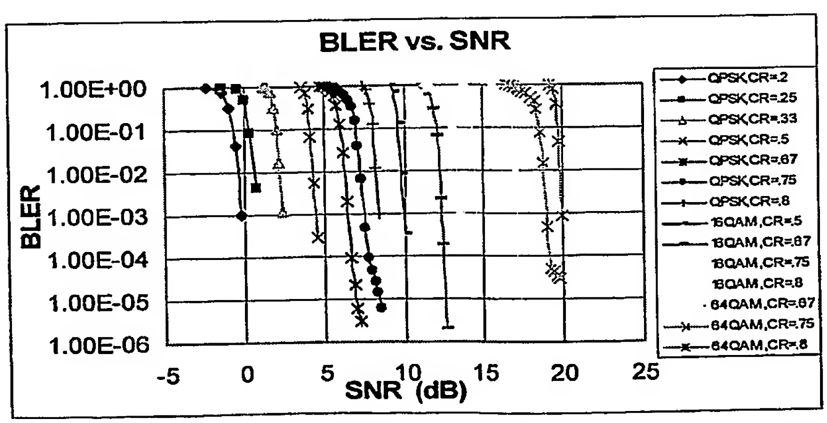

Figure 10: Block error rate (BLER) versus SNR with bandwidth of 6MHz.

1.

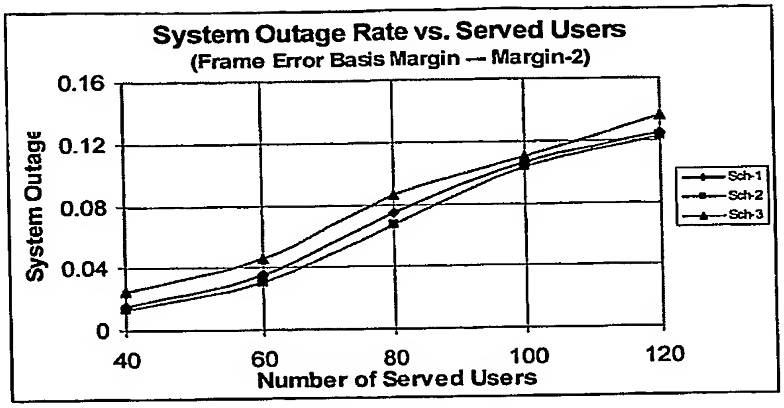

Figure 11: System outage versus the number of served users when using frame error based margin (margin-2) for AMC for various specified schedulers.

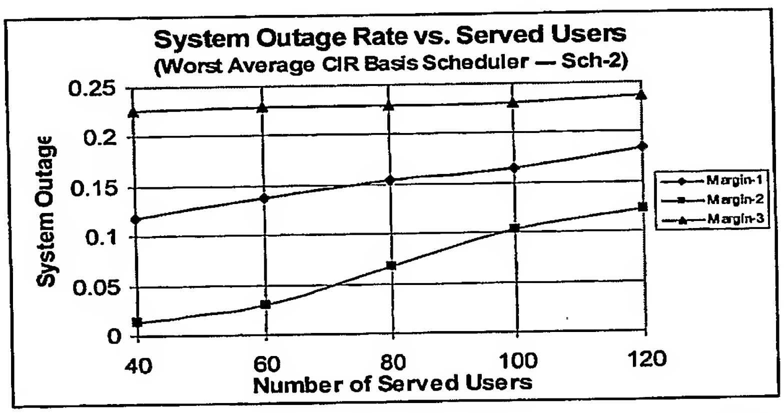

Figure 12: System outage versus the number of served users when using the worst average CIR based scheduler (scheduler-2) for various specified margins.

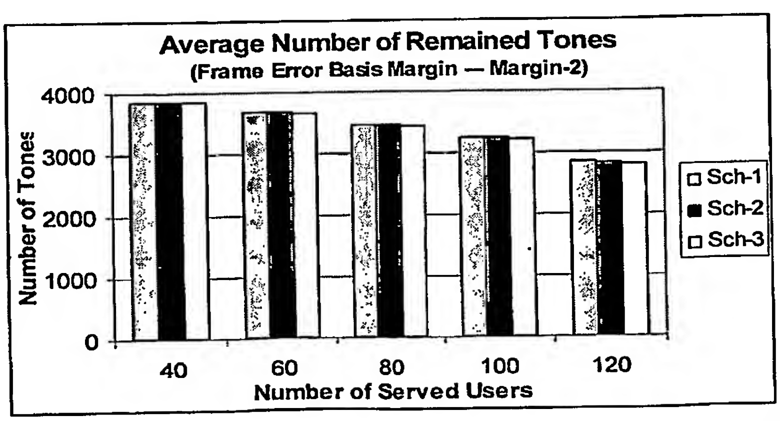

Figure 13: Average number of remained OFDM tones versus number of served users when using frame error based margin (margin-2) for AMC for various specified schedulers.

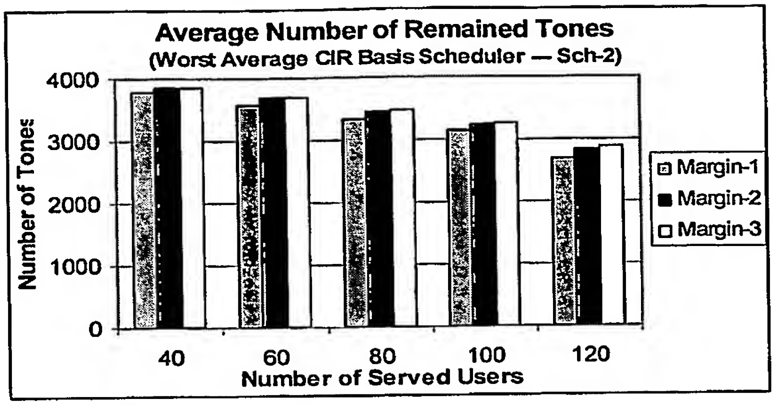

Figure 14: Average number of remained OFDM tones versus number of served users when using the worst average CIR based scheduler (scheduler-2) for various specified margins.

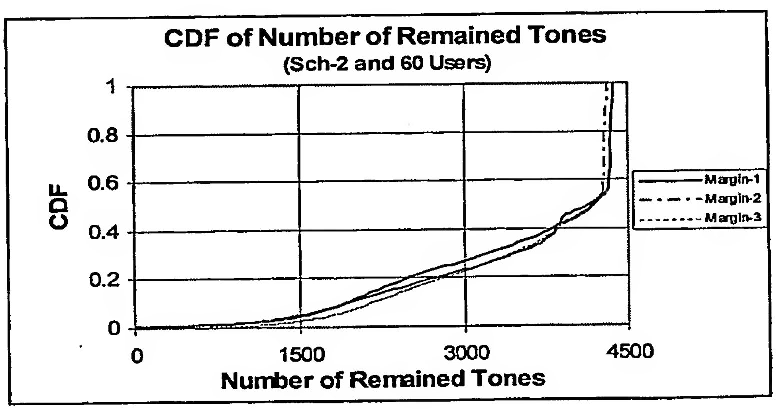

Figure 15: CDF of the number of remained OFDM tones when using the worst average CIR based scheduler (scheduler-2) for various specified margins.

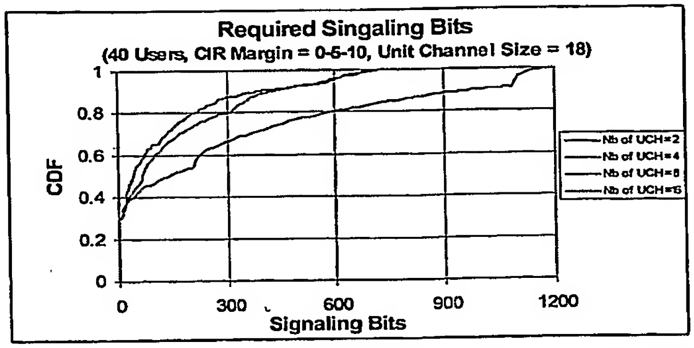

Figure 16: CDF of signaling bits for indicating the unit channel address.

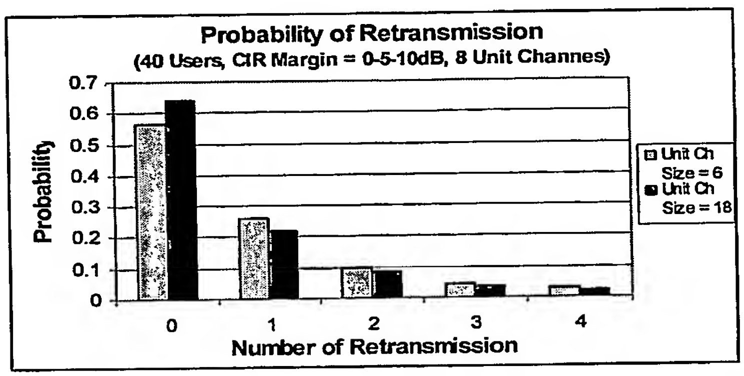

Figure 17: Probability of voice packet retransmission with 40 users, CIR margin of 0-5-10dB and 8 unit channels.

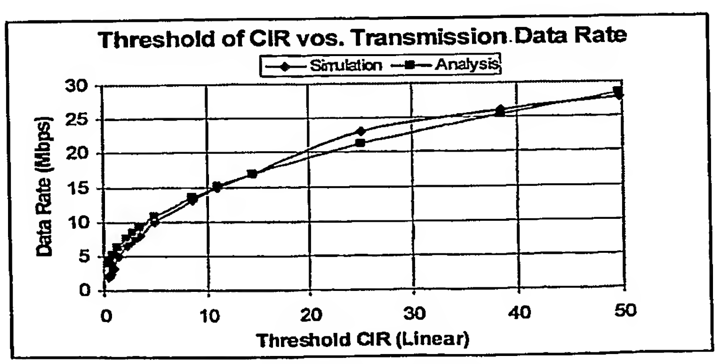

Fig 18: Approximation of MCS threshold based on 1% BLER.

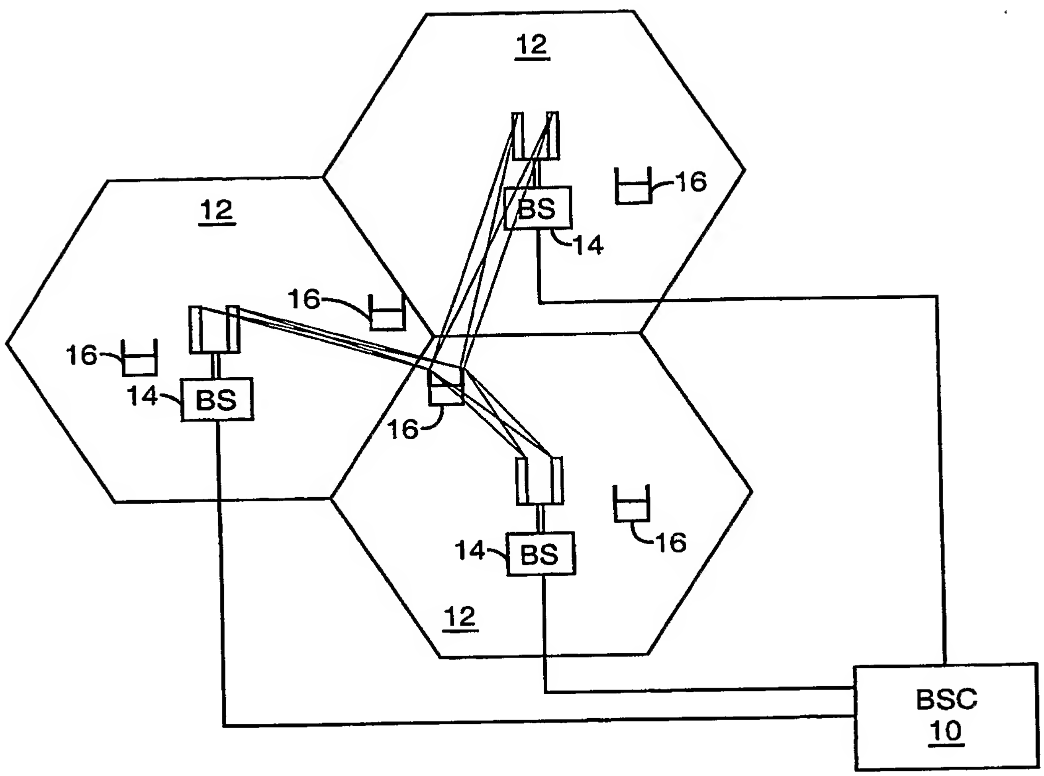

FIG. 19

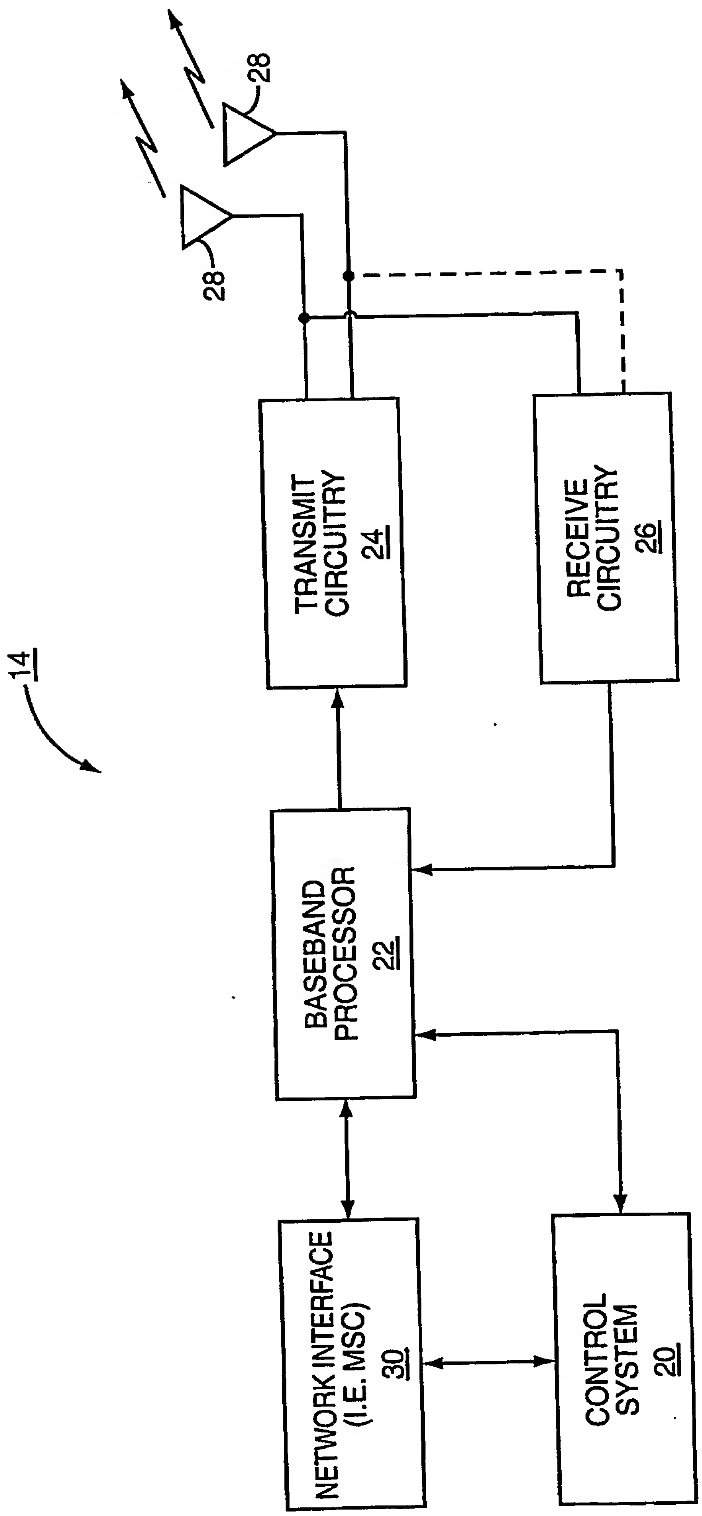
14
28
28
TRANSMIT
CIRCUITRY
24
RECEIVE
CIRCUITRY
26
BASEBAND
PROCESSOR
22
NETWORK INTERFACE
(I.E. MSC)
30
CONTROL
SYSTEM
20

FIG. 20

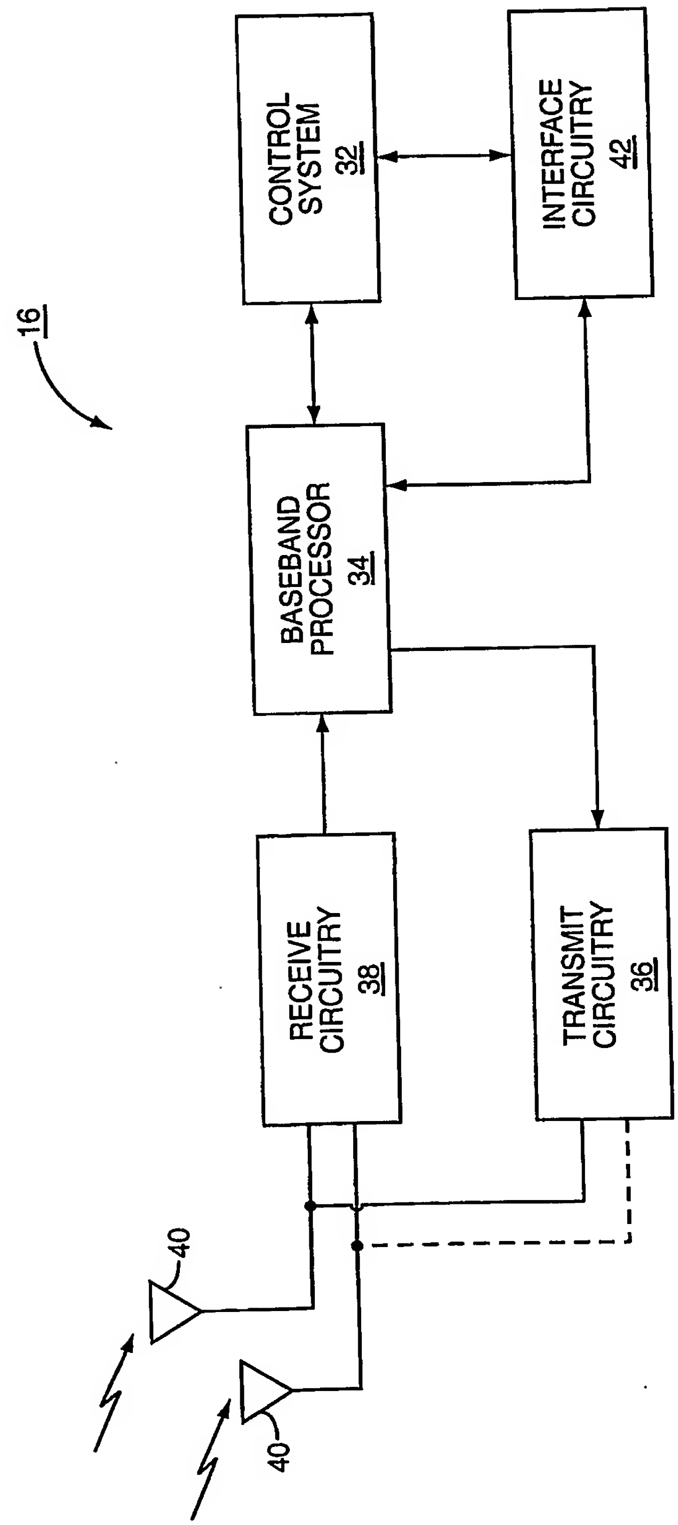
16
CONTROL SYSTEM 32
INTERFACE CIRCUITRY 42
BASEBAND PROCESSOR 34
RECEIVE CIRCUITRY 38
TRANSMIT CIRCUITRY 36
40
40

FIG. 21

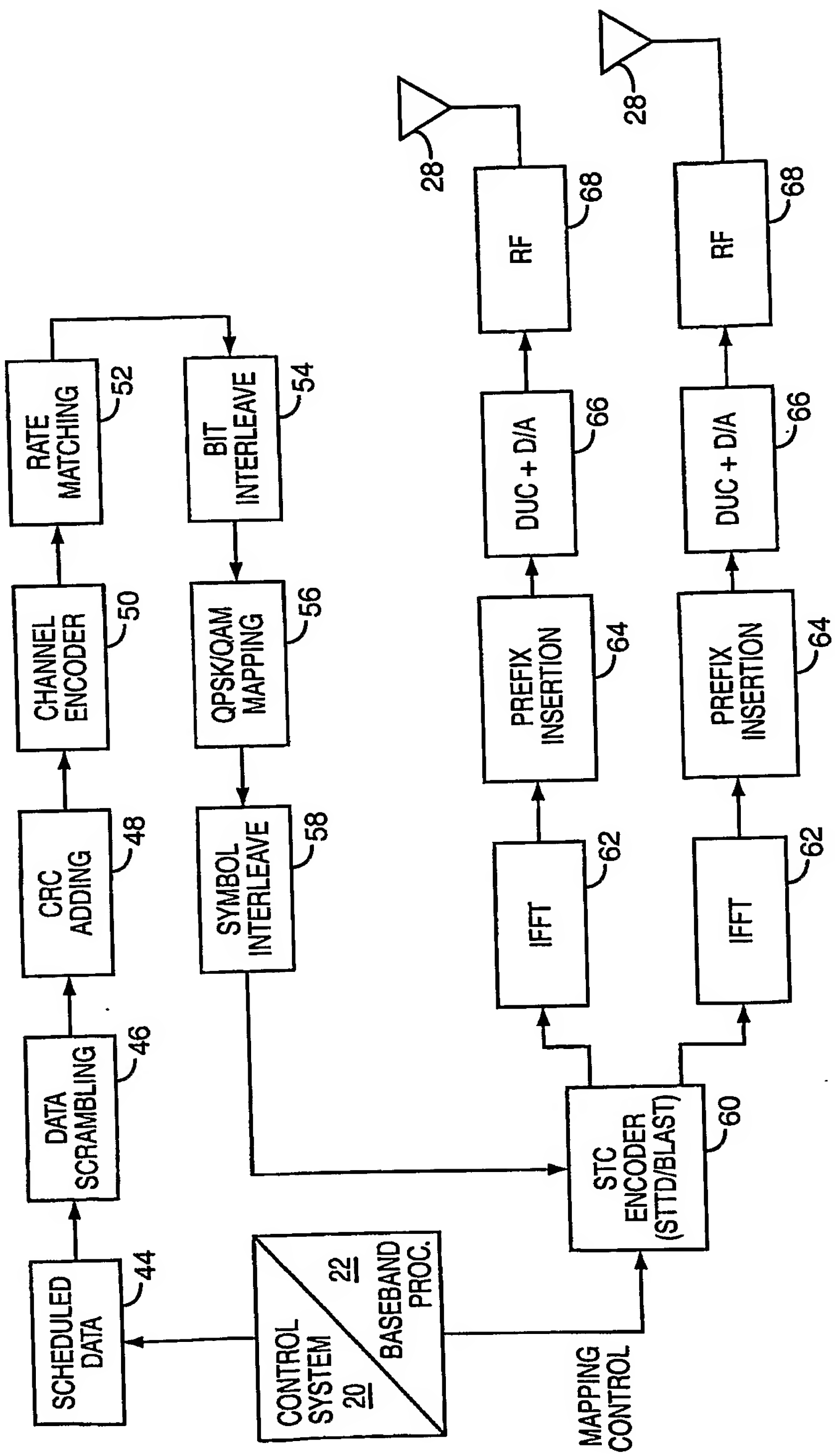
SCHEDULED DATA
44
DATA SCRAMBLING
46
CRC ADDING
48
CHANNEL ENCODER
50
RATE MATCHING
52
BIT INTERLEAVE
54
QPSK/QAM MAPPING
56
SYMBOL INTERLEAVE
58
CONTROL SYSTEM 20
BASEBAND PROC. 22
MAPPING CONTROL
STC ENCODER (STTD/BLAST)
60
IFFT
62
PREFIX INSERTION
64
DUC + D/A
66
RF
68
28

FIG. 22

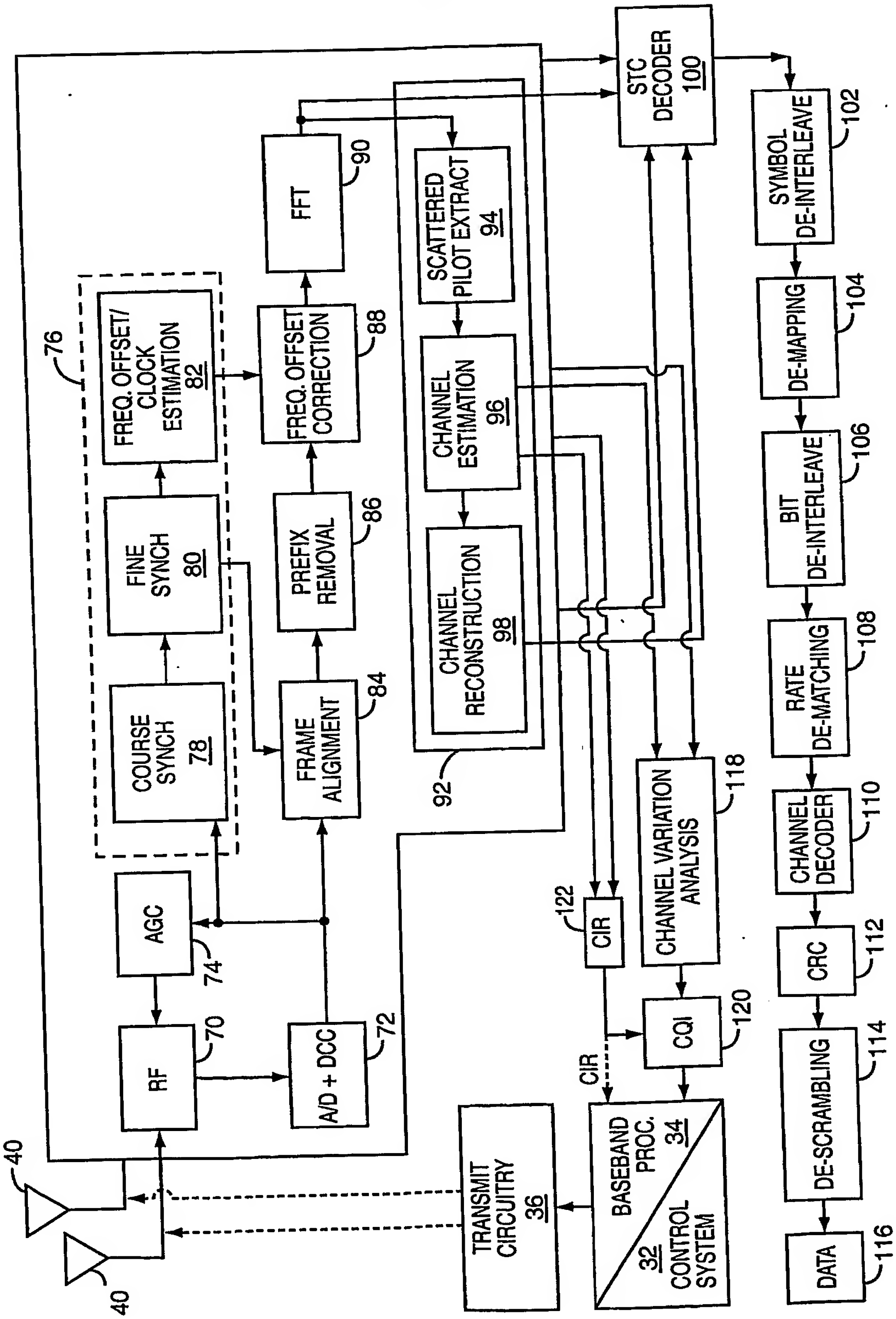
40
40
RF
70
A/D + DCC
72
AGC
74
76
COURSE SYNCH 78
FINE SYNCH 80
FREQ. OFFSET/ CLOCK ESTIMATION 82
FRAME ALIGNMENT
84
PREFIX REMOVAL
86
FREQ. OFFSET CORRECTION
88
FFT
90
92
SCATTERED PILOT EXTRACT 94
CHANNEL ESTIMATION 96
CHANNEL RECONSTRUCTION 98
STC DECODER 100
SYMBOL DE-INTERLEAVE
102
DE-MAPPING
104
BIT DE-INTERLEAVE
106
RATE DE-MATCHING
108
CHANNEL DECODER
110
CRC
112
DE-SCRAMBLING
114
DATA
116
CHANNEL VARIATION ANALYSIS
118
CQI
120
CIR
122
CIR
BASEBAND PROC. 34
32 CONTROL SYSTEM
TRANSMIT CIRCUITRY 36

FIG. 23